SONY

Digital Sonic Modulator

Operating Instructions
Mode d'emploi

DPS-M7

Warning

FOR CUSTOMERS IN THE UNITED STATES

Owner's Record

The model and serial numbers are located at the rear. Record these numbers in the space provided below. Refer to these numbers whenever you call upon your Sony dealer regarding this product.

Model No. DPS-M7 Serial No. ____________

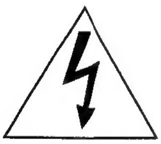
WARNING

To prevent fire or shock hazard, do not expose the unit to rain or moisture.

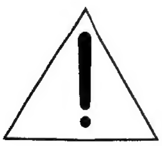
CAUTION

RISK OF ELECTRIC SHOCK
DO NOT OPEN

CAUTION : TO REDUCE THE RISK OF ELECTRIC SHOCK,
DO NOT REMOVE COVER (OR BACK).
NO USER-SERVICEABLE PARTS INSIDE.
REFER SERVICING TO QUALIFIED SERVICE PERSONNEL.



This symbol is intended to alert the user to the presence of uninsulated "dangerous voltage" within the product's enclosure that may be of sufficient magnitude to constitute a risk of electric shock to persons.



This symbol is intended to alert the user to the presence of important operating and maintenance (servicing) instructions in the literature accompanying the appliance.

For detailed safety precautions, see the leaflet "IMPORTANT SAFEGUARD".

WARNING

This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

- Reorient or relocate the receiving antenna.
- Increase the separation between the equipment and receiver.
- Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.
- Consult the dealer or an experienced radio/TV technician for help.

You are cautioned that any changes or modifications not expressly approved in this manual could void your authority to operate this equipment.

- The shield interface cable recommended in this manual must be used with this equipment in order to comply with the limits for a computing device pursuant to Subpart J of Part 15 of FCC Rules.

FOR CUSTOMERS IN CANADA

CAUTION

TO PREVENT ELECTRIC SHOCK, DO NOT USE THIS POLARIZED AC PLUG WITH AN EXTENSION CORD, RECEPTACLE OR OTHER OUTLET UNLESS THE BLADES CAN BE FULLY INSERTED TO PREVENT BLADE EXPOSURE.

This apparatus complies with the Class B limits for radio noise emission set out in Radio Interference Regulations.

Precautions

FOR CUSTOMERS IN THE UNITED KINGDOM

WARNING
THIS APPARATUS MUST BE EARTHED

IMPORTANT
The wires in this mains lead are coloured in accordance with the following code.

Green-and -yellow:	Earth
Blue:	Neutral
Brown:	Live

As the colours of the wires in the mains lead of this apparatus may not correspond with the coloured markings identifying the terminals in your plug proceed as follows:
The wire which is coloured green-and-yellow must be connected to the terminal in the plug which is marked by the letter E or by the safety earth symbol ⏚ or coloured green or green-and-yellow. The wire which is coloured blue must be connected to the terminal which is marked with the letter N or coloured black. The wire which is coloured brown must be connected to the terminal which is marked with the letter L or coloured red.

On Safety
- To avoid electrical shock, do not open the cabinet. Refer servicing to qualified personnel only.
- Before connecting the unit to the power source, check to confirm that the operating voltage of your unit is the same as the local power line voltage. The operating voltage is indicated on the nameplate on the left side of the unit.
- Should anything fall into the cabinet, unplug the unit and have it checked by qualified personnel before operating it any further.
- The unit is not disconnected from the mains (AC power source) as long as it is connected to the mains outlet, even if the unit itself has been turned off.

On Installation
- Allow adequate air circulation to prevent internal heat build-up.
- Do not place the unit on a surface (rugs, blankets, etc.) or near materials (curtains, draperies, etc.) that may block the ventilation holes.
- Do not install the unit near heat sources such as radiators or air ducts or in a place subject to direct sunlight, excessive dust, mechanical vibration or shock.
- The unit is designed for operation in a horizontal position. Do not install it in an inclined position.
- Keep the unit away from equipment with strong magnets, such as microwave ovens or large loudspeakers.
- Do not place any heavy object on the unit.

On Operation
- When the unit is not in use, turn the power off to conserve energy and to extend its life.

On Cleaning
- Clean the cabinet, panel and controls with a dry soft cloth, or a soft cloth slightly moistened with a mild detergent solution.
- Do not use any type of solvents, such as alcohol or benzine, which might damage the finish.

On Repacking
- Do not throw away the carton and packing materials. They make an ideal container to transport the unit.

If you have any questions about the unit, contact your Sony service facility.

CAUTION!
Danger of explosion if battery is incorrectly replaced.
Replace only with the same or equivalent type recommended by the equipment manufacturer.
Discard used batteries according to manufacturer's instructions.

Table of Contents

Overview of the DPS-M7

The DPS-M7 is a digital sonic modulator developed with the wealth of digital and audio technology accumulated over the years by Sony, innovater of the highly acclaimed Digital Reverberator DRE-2000 and MU-R201.

Quality-conscious design with high-performance A/D and D/A converters
The DPS-M7 converts an incoming analog signal to a digital signal and outputs the signal again after passing it through various effect processes and reconverting it into an analog signal. The determining factor for the sound quality is the conversion mechanism that adopts the 18-bit oversampling stereo A/D converter and the 49.152 MHz clock advanced pulse D/A converter, which results in highly accurate effects with little deterioration of quality.

User-friendly and comfortable operation
The large size backlit LCD of 40 characters by 2 lines enables smooth operation while viewing the operating condition in real time. Since the LCD also has an on-line manual function (in English), information necessary for operation can be displayed.

Abundant preset memories
The DPS-M7, in its preset memory, has a hundred variations of effects created by musicians, sound mixers and acoustic engineers around the world . This will therefore enable you to select and replay the desired effects for a particular purpose immediately.

Creation of any kind of sound
The EDIT function allows you to change the preset effects or create original effects. Aside from the present preset memory of 100 effects, the DPS-M7 also has a user memory in which up to 256 effects can be freely saved. Using this memory will enable more varicolored play effects.

Wide range of effects
To obtain various effects, the DPS-M7 processes signals with seven blocks consisting mainly of the modulation block, plus the input block, pre-effect blocks 1 and 2, post effect block, envelope block and output block.
One of the 20 algorithms available in the modulation block can be used. One of the 5 algorithms available in pre-effect 1 and 2 blocks can be used. One of the 4 algorithms in post-effect block and one of the 3 algorithms in the envelope blocks can be used. (Algorithms "OFF" are excluded.)
Variegated effects matching the input source can be obtained by combinations of the seven blocks and combinations of the algorithms in the blocks.

Remote control is possible
Remote control of the panel operation is possible by means of the separately available remote controller (RM-DPS7).

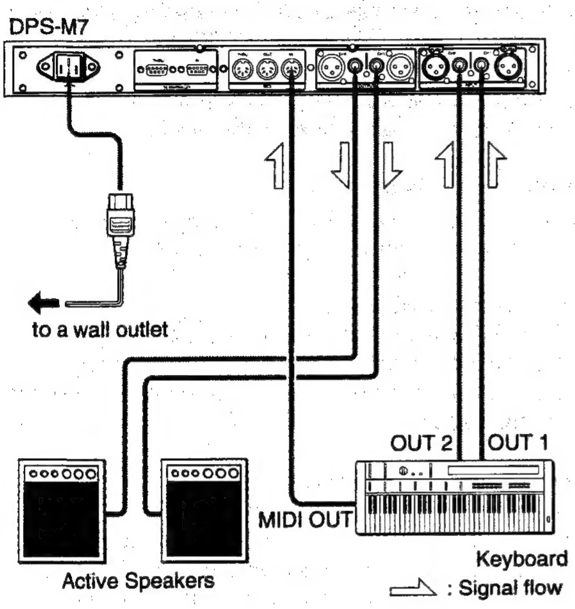
Two I/O terminal systems
The DPS-M7 is equipped with XLR connectors (balanced type) and phone jacks to which musical instruments, recording equipment and PA (public address) equipment can be connected.

Linkage with MIDI equipment
Since the DPS-M7 is equipped with MIDI functions, memory numbers of this unit can be selected with program change signals of the MIDI device such as a keyboard. Moreover, since effect level, etc. can be controlled by key touch and control change signals, the unit is highly effective as an effector of digital musical instruments. Automatic performance is also possible by controlling with computers having the MIDI interface and with a MIDI sequencer.

Hooking Up a System

Turn all the power off before making connections, and connect the AC power cord last.

Fundamental Connections as an Effector

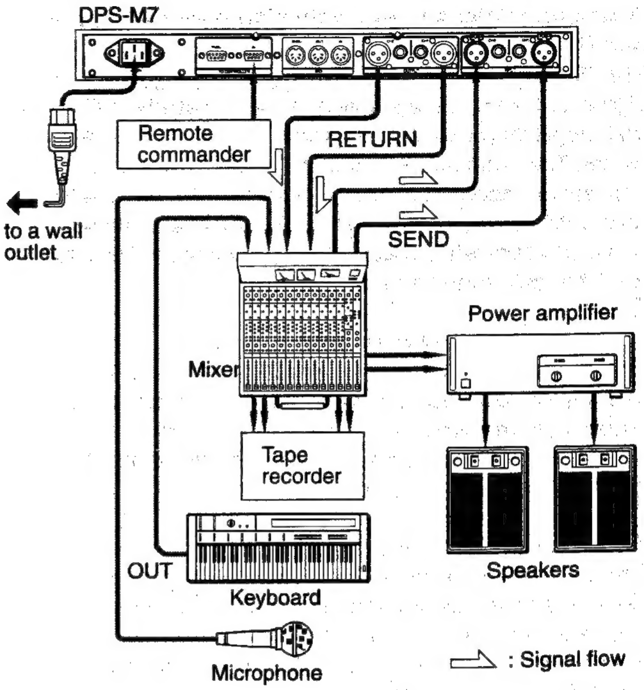


1. Connect a keyboard to the INPUT jacks, or the MIDI IN connector.
2. Connect active speakers to the OUTPUT jacks.
3. Insert the AC power cord firmly into the AC IN jacks.
4. Connect the AC power cord to a wall outlet.

For the model equipped with a voltage selector
Check to confirm that the voltage selector is set to the local power line voltage. If not, set the selector to the correct position before connecting the AC power cord to a wall outlet.

Notes:
- Be sure to insert the plugs firmly into the jacks. Loose connection may cause hum and noise.
- Leave a little slack in the connecting cord to allow for inadvertent shock or vibration.
- Connections with some equipment of which the ouput capacity is very high may result in sound distortion. When this happens, turn the INPUT control to lower the input level of the equipment connected to the DPS-M7.

Fundamental Connections for Recording

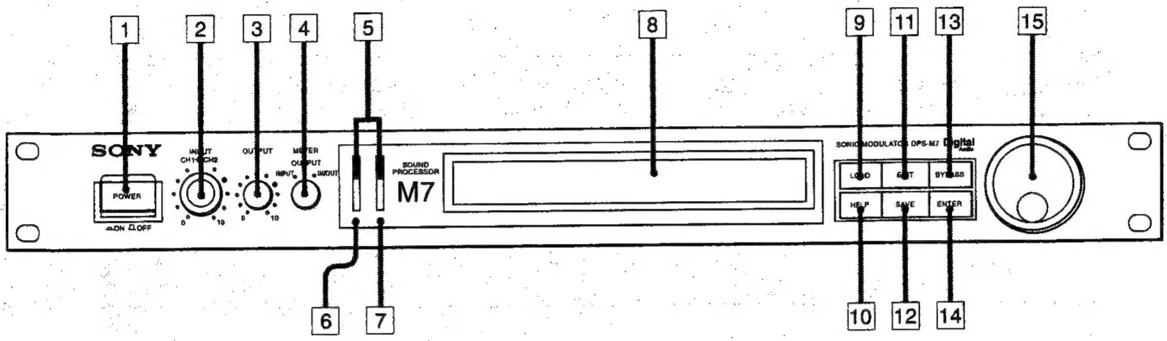


Notes:
- If there is only one channel for the input signal, input to INPUT CH1 and set the input mode in the system block to "mono". This will have the same effect as inputting the same signal into INPUT CH1 and INPUT CH2 with the input mode set to "stereo".
- Always input signals with a reference level of +4 dBs through an XLR-3-31 type connector.
- The reference level of a phone jack is fixed at −10 dBs. Therefore, if the maximum input level of the input signal exceeds +10 dBs, distortion will occur since the amplifier preceding the input volume control clips the signal.
- An optional remote controller RM-DPS7 can be connected to the TO CONTROLLER IN connector to remotely control this unit.

Identifying the Parts

To be continued ►

Front Panel



1 POWER Switch
Turns the unit on and off. When the power is on, the backlight in the display window illuminates and the last indication appears. For a few seconds after turning on the power, the sound being input will be output directly since the bypass function works.

2 INPUT control
Adjusts the input level of individual channels.
The outer control is for channel 1 and the inner control is for channel 2. Since the controls are linked, turn one control while holding the other for adjustment of only one channel. Gain will become 0 dB if this control is turned up to the two o'clock position (largest point on the scale).

3 OUTPUT control
Adjusts the output level. Gain will become 0 dB if this control is turned up to the two o'clock position (largest point on the scale).

4 METER switch
Switches signals to be indicated on the level meter. If the switch is set to INPUT, the input signal level of each channel to the A/D converter will be indicated individually and, if set to OUTPUT, the output signal level of each channel from the D/A converter will be indicated individually. When set to IN/OUT, the channel 1 signal level being input to the A/D converter will be indicated on CH1 of the level meter and the level of the channel 2 signal output from the D/A converter will be indicated on CH2 of the level meter.

5 Level meter
Indicates the signal level. Adjust the INPUT control so 0 dB lights when a reference level signal is input. A 20 dB head room will be available when 0 dB lights. "OVER" will light if a signal exceeding the head room is input. The level meter remains inactive when the BYPASS button is pressed.

6 MIDI indicator
Lights when the MIDI program change signal or the control change signal, etc. is received.

7 REMOTE indicator
Lights when a signal is received from an optionally available remote controller (RM-DPS7).

8 Display window
Memory names, parameter values and messages accessed are displayed on an LCD display of 40 characters by 2 lines. Displayed indications can be easily read in dark halls and studios due to the backlighting.

9 LOAD button
Press to access the memory.

10 HELP button
Press to display various information required for operation. Message will be displayed if this button is pressed.

11 EDIT button
Press to change parameter values of the memory.

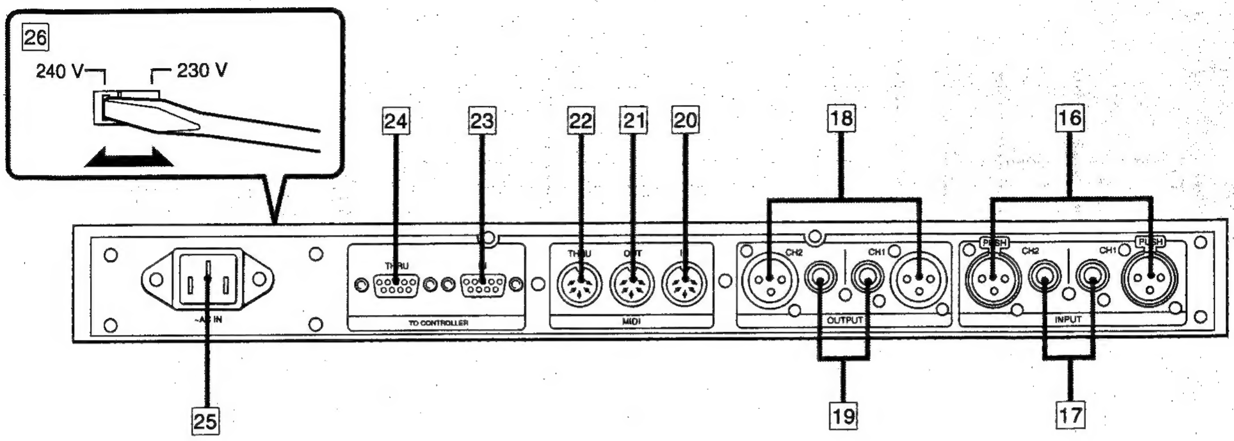
12 SAVE button
Press when storing original effects created by changing parameter values in the user memory.

13 BYPASS button
Press when outputting input signals directly.

14 ENTER button
Press after selecting and setting parameters.

15 Operating dial
Selects preset numbers and/or sets parameters.

Rear Panel



16 INPUT CH1/CH2 terminal (XLR-3-31 connector)
Balanced-type terminals for input of ch1 and ch2.

17 INPUT CH1/CH2 terminal (Phone jacks)
Phone jacks for input of ch1 and ch2.

18 OUTPUT CH1/CH2 terminal (XLR-3-32 connector)
Balanced-type terminals for output of ch1 and ch2.

19 OUTPUT CH1/CH2 terminal (Phone jack)
Phone jacks for output of ch1 and ch2.

* When devices are connected to both XLR connectors and phone jacks, the device connected to the phone jacks will have priority.

20 MIDI IN terminal (DIN 5-pin)
Input terminal for the MIDI signal. For connection to the MIDI OUT (or THRU) terminal of another MIDI device by means of a commercially available MIDI cable.

21 MIDI OUT terminal (DIN 5-pin)
Outputs the MIDI signal generated in this unit.

22 MIDI THRU terminal (DIN 5-pin)
Outputs MIDI signals input from the MIDI IN terminal as is, and can be connected to the MIDI IN terminal of a MIDI device with a commercially available MIDI cable.

23 TO CONTROLLER IN terminal (D-Sub 9-pin)
Terminal to which the remote controller RM-DPS7 (not supplied) is connected to permit remote control of panel operation of the DPS-M7.

24 TO CONTROLLER THRU terminal (D-Sub 9-pin)
Outputs directly the remote controller signals input from the TO CONTROLLER IN terminal. Connect to the TO CONTROLLER IN terminal of other effectors in the DPS series.

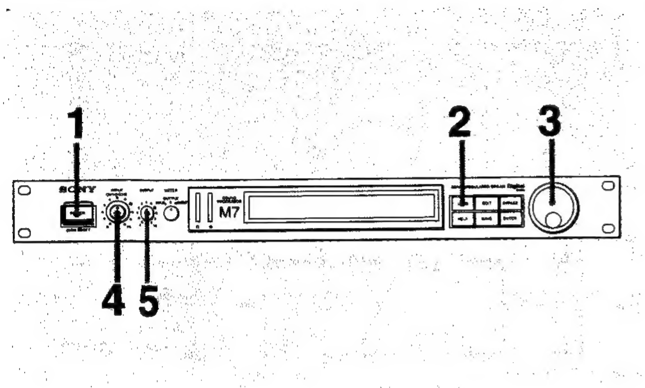
25 AC IN terminal
Use the supplied AC power cord and connect it to an AC power outlet.

26 VOLTAGE SELECTOR
(only for UK and European model)
Set the voltage selector to the correct position before connecting the AC power cord to a power outlet.

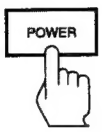
Let's Try to Operate Your DPS-M7

The DPS-M7 has a hundred effects stored in its preset memory. Let's therefore listen to these effects one by one, referring to "Hooking Up a System" (page 6) and "Preset Memory List" (separate volume).

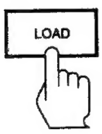
Selecting a Preset Memory

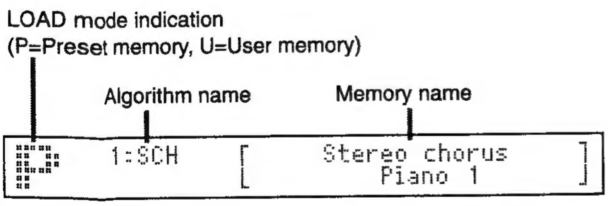


1. Turn on the power.



2. Press the LOAD button.

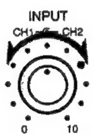




3. Turn the operating dial and select the desired preset number (P1-P100, U1-U256).

- For the contents of the preset memories (from P1 to P100), refer to the "Preset Memory List" (separate volume).
- Only stored preset numbers (from U1 toU256) can be selected. (See page 63)

4. Turn the INPUT control to adjust the input level .



5. Turn the OUTPUT control to adjust the output level.

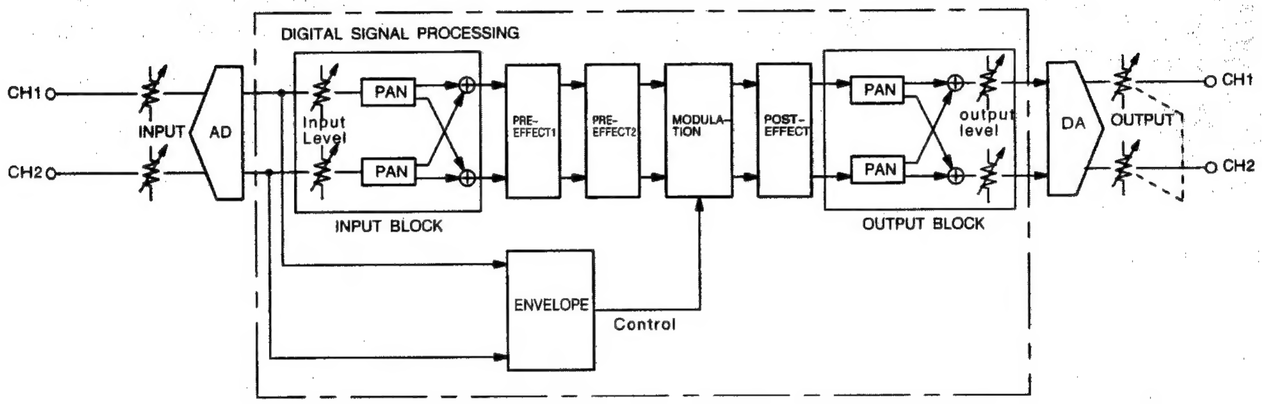
Before turning on the connected devices
Lower the volume of each device to prevent unexpected loud sounds.

To output the input signals directly
Press the BYPASS button when outputting the input signal directly. The input signal will then be output directly. To release the bypass function, press the button once again.

Overview of the Signal Processing Blocks

The signal digitalized by the A/D converter is processed sequentially in the input, pre-effect 1, pre-effect 2, modulation, post-effect and output blocks, and the results are sent to the D/A converter.

General Block Diagram



Note
When using the bypass function, signals input to ch1 and ch2 bypass the digital circuit and are output directly to the output terminal. When switching off the unit, the bypass function is automatically applied.

Since the digital signal processing section has a 12 dB margin over the full-bit output signal from the A/D converter, the signal level raised within 12 dB by EQ (equalizer) parameters in the digital processing section can be regulated simply by adjusting the output level to prevent the clipping. If the signal is supposed to be raised more than 12 dB, lower the input level.

The analog signal processing section has a gain of 10 dB for each input and output. Adjust the input and output levels with the INPUT and OUTPUT controls to make the I/O level suitable for the equipment connected to the DPS-M7. Setting the controls to the two o'clock position (largest point on the scale) produces a gain of 0 dB approximately.

Algorithm 0

The pre-effect 1, pre-effect 2, modulation, and post-effect blocks contain an algorithm called "Algorithm 0." If set to "Algorithm 0," that block will be passed through. When desiring to set a block in a user memory to "Algorithm 0," load and save the pertinent block with the B.LOAD function (see page 57) from a preset memory or another user memory containing "Algorithm 0" as for other algorithms.

What can be done in each block?

Input block: The first block to receive digital signals. See page 12 for details.

Pre-effect 1 & 2 blocks: Pre-processing block closely related to the post effect block. See page 13 to 17 for details.

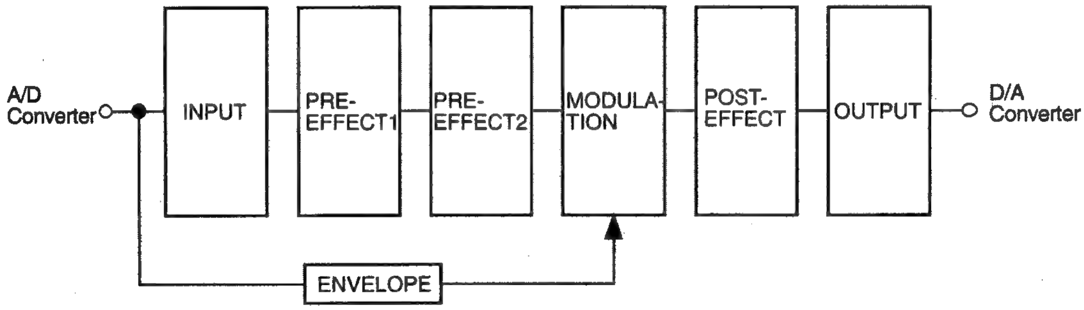
Modulation block: The most important block in creating digital sound effects. See pages 18 to 48 for details.

Post-effect block: Post-processing block closely related to the pre-effect blocks. See page 49 for details.

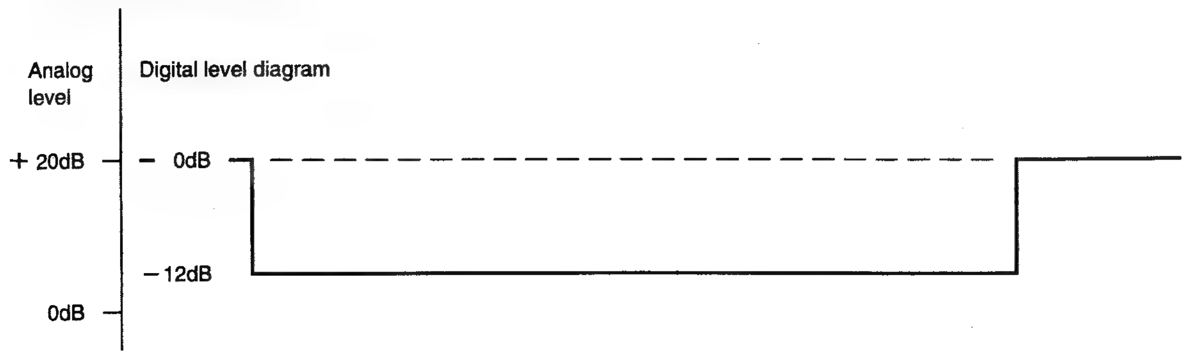
Envelope block: Used for real time control of the modulation block. Basic waveforms are formed in this block. See page 50 to 52 for details.

Output block: The final block to send signals to the D/A converter. See page 53 for detail.

Signal flow diagram



Analog level-digital level diagram

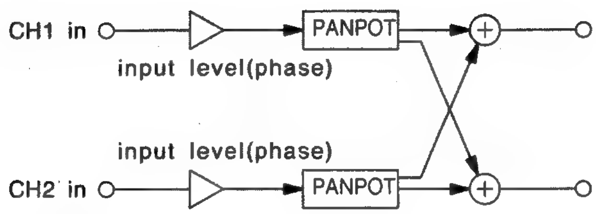


Input Block

This block receives signals from the A/D converter and gives them level, phase and panpot processing.

Parameter	MIN and MAX
input level (ch1, ch2)	0 – 100%
input phase (ch1, ch2)	normal/inverse
input panpot (ch1, ch2)	0 – 100%

Note:
If the "input panpot" parameter is set to 0%, signals pass through the panpot processing, and if it is set to 100%, input signals for ch1 is output to ch2 and vice versa.



Pre-effect 1 and 2 Blocks

To be continued ►

This block receives signals from the input block, processes them successively in pre-effect 1 and pre-effect 2 blocks and outputs them to the modulation block. Signal processing in these blocks uses 5 different algorithms (excluding Algorithm 0) according to the preset memory.

When editing a preset memory, first confirm the type of algorithm used in the preset memory. Parameters also differ according to the algorithm.

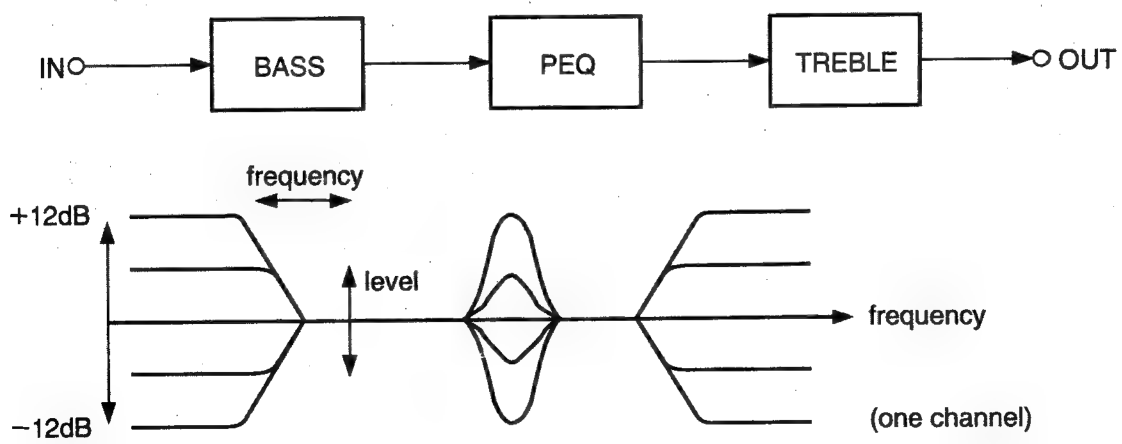
Algorithm 0 — Effect Off — OFF

Pre-effect 1 and 2 block will be passed though so that no effect can be obtained in these blocks.

Algorithm 1 — Stereo Equalizer — SEQ

This is a dual channel digital equalizer and is composed of 3 band equalization (bass, treble, peaking) independently operated for each channel.

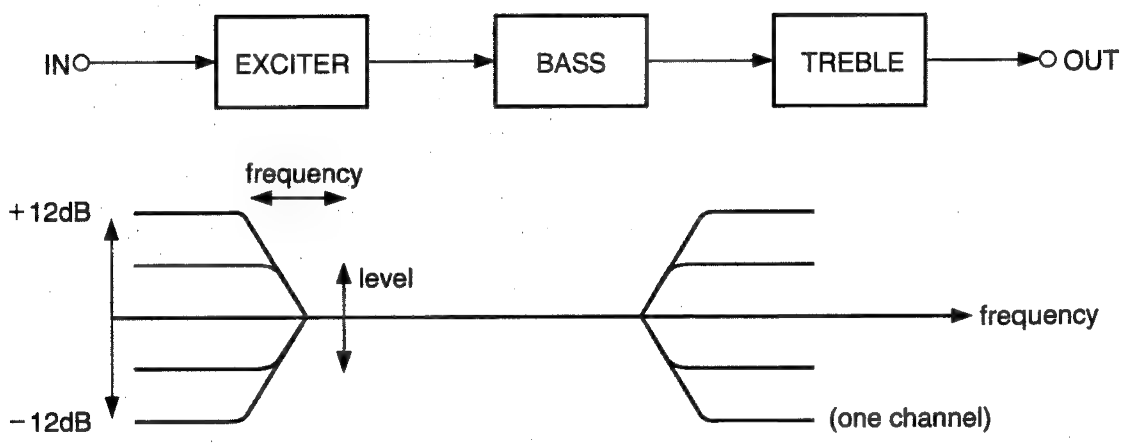
Parameter	**MIN and MAX**
stereo EQ on/off	on /off
bass frequency (ch1, ch2) bass level (ch1, ch2)	16Hz – 6.3kHz –12 – +12dB
treble frequency (ch1, ch2) treble level (ch1, ch2)	400Hz – 20.0kHz –12 – +12dB
PEQ frequency (ch1, ch2) PEQ level (ch1, ch2) PEQ q (ch1, ch2)	63Hz – 20.0kHz –12 – +12dB 0.267/0.667/1.414/2.145 4.319/8.651/17.31/34.62



Algorithm 2 Stereo Exciter + Stereo EQ SXE

This is a dual channel digital equalizer with a dual channel exciter. The equalizer consists of 2 band equalization (bass, treble) independently operated for each channel. The exciter clarifies the profile of the incoming signal, modulates the sound itself and has the effect of creating stressed sound.

Parameter	MIN and MAX
stereo exciter + EQ on/off	on/off
exciter level (ch1, ch2)	–100 – +100 %
bass frequency (ch1, ch2) bass level (ch1, ch2)	16Hz – 6.3kHz –12 – +12dB
treble frequency (ch1, ch2) treble level (ch1, ch2)	400Hz – 20.0kHz –12 – +12dB



To be continued ►

Algorithm 3 Dynamic Exciter DEX

This algorithm provides clear start-up of sound and also a firm sound profile. This will be effective when applied to percussive sounds since the degree of excitation increases as the difference in volume increases. The time during which the exciter effect is held is set by "release time."

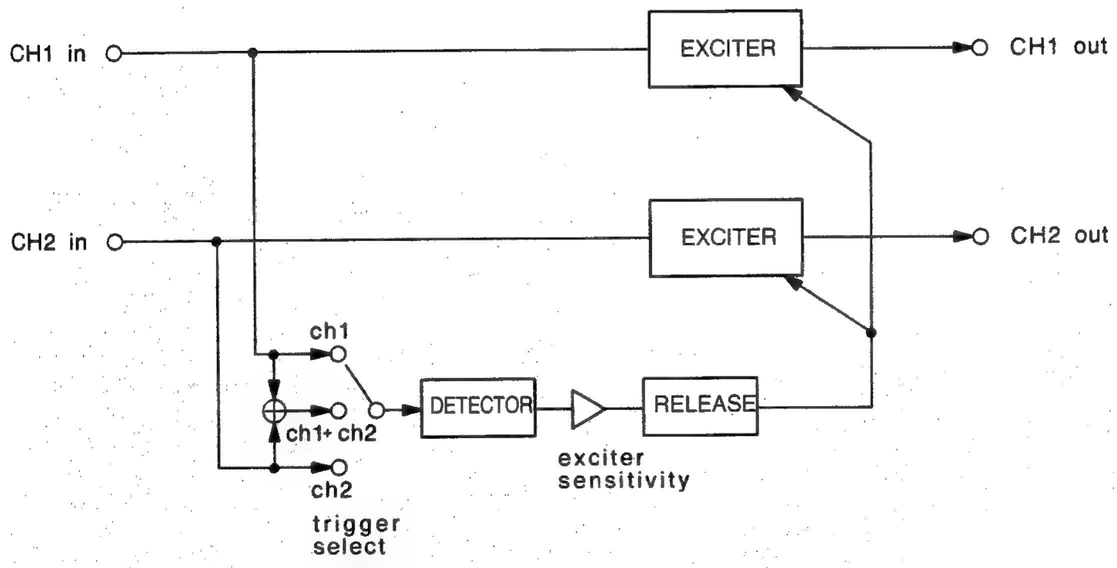
Parameter	MIN and MAX
dynamic exciter on/off	on/off
trigger select	ch1/ch2/ch1+ch2
exciter sensitivity	0 – 100%
exciter type	1 – 4
release time	1 – 5000msec

Parameter description

exciter sensitivity
Adjusts the output level of the detector and sets the sensitivity of the exciter.

exciter type
Selects the frequency characteristics of the exciter.

release time
Sets the holding time of exciter effects by difference in volume.

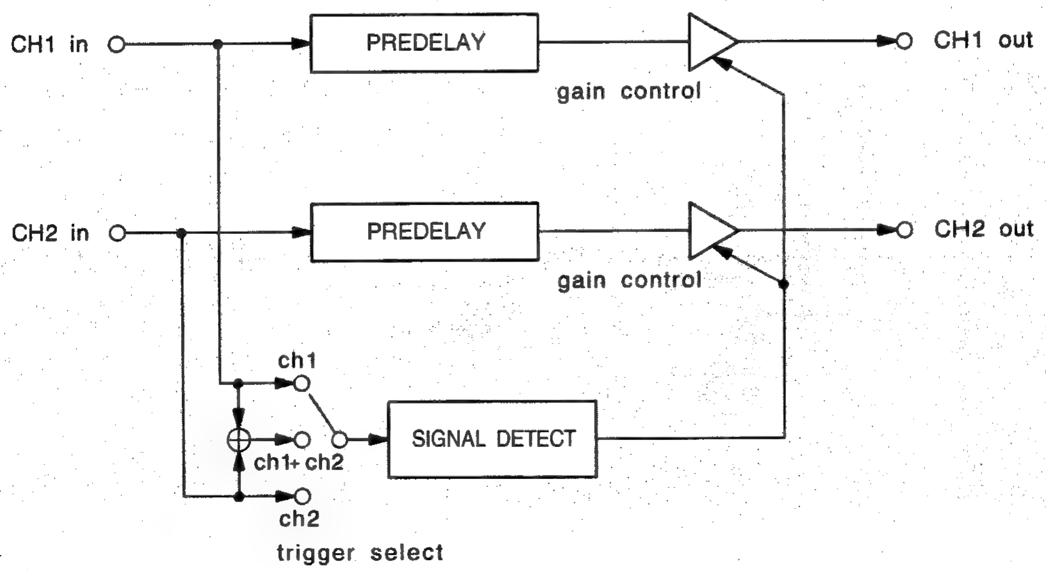


Algorithm 4 Gate GTE

This is a dual channel gate. The function of a gate is to turn the output signal on and off according to the level of the input signal. The parameters in this algorithm will enable you to output attacking sounds only or eliminate noise when there is no signal.
Moreover, since the time constant for the gate on/off can be changed, effective sounds can be created by changing the envelope of the sound during attacking or releasing.
Predelay in the main line is to adjust the time required for signal detection. For example, predelay can be applied to the original sound to create an effect as if it had been given gate processing.

Parameter	MIN and MAX
gate on/off	on/off
trigger select	ch1/ch2/ch1+ch2
attack time release time	0 – 500msec 1 – 5000msec
threshold level hysterisis level	0 – 100% 0 – 100%
predelay time (ch1, ch2)	0 – 9words

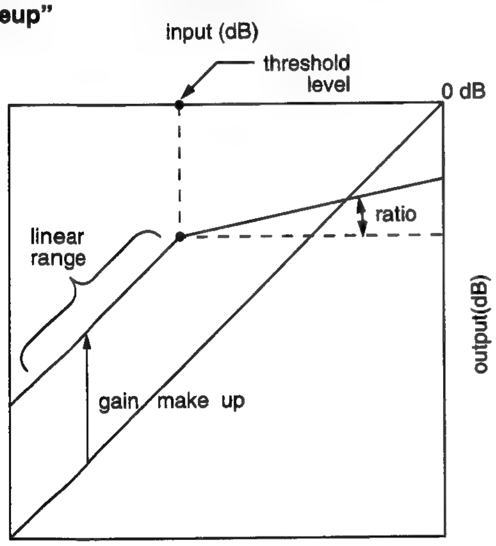
Parameter description
threshold level
Sets the level at which the gate turns on.
hysterisis level
Sets the level at which the gate turns off. "hysteresis level" is expressed in proportion to "threshold level". For example, gate on and gate off become the same level when "hysteresis level" = 0; gate off occurs at one half the level of gate on when "hysteresis level" = 50%.



Algorithm 5 Compressor CMP

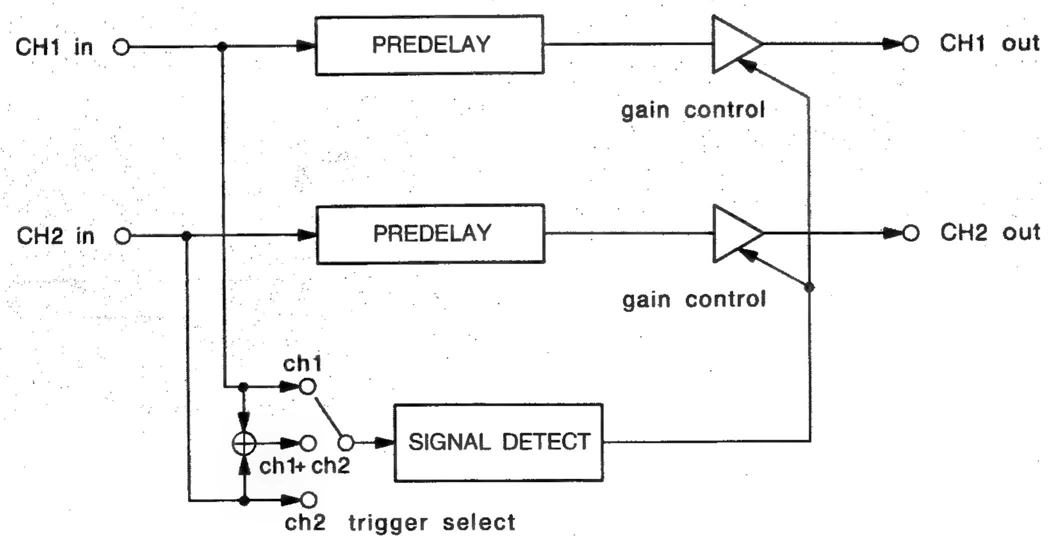
This is a dual channel compressor. This is used to produce the feeling of power and suppress excessive input by smoothing the large sound source in the dynamic range. Predelay in the main line is to adjust the time required for signal detection. For example, predelay can be applied to the original sound to create an effect as if it had been given a compressor processing. When the compressor output level becomes low because of lowering of the "threshold level" or increasing of the "ratio," adjust it with "gain makeup."

• **Relation among "threshold level," "ratio" and "gain makeup"**



Parameter	MIN and MAX
compressor on/off	on/off
trigger select	ch1/ch2/ch1+ch2
attack time release time	0 – 500msec 1 – 5000msec
threshold level	0 – 100%
ratio	1 : 1 – 128 : 1
gain makeup	0 – +24dB
predelay time (ch1, ch2)	0 – 9words

Note:
Clipping may occur depending on the set "threshold level," "ratio" or "attack time" value if the "gain makeup" value is set too high.



Modulation Block

This block processes output of pre-effect 2 block and outputs to the post-effect block. Signal processing in this block uses 20 different algorithms (excluding Algorithm 0) according to the preset memory.

When editing a preset memory, first confirm the type of algorithm used in the preset memory. Parameters also differ according to the algorithm.

Dynamic Modulation Function

The dynamic modulation function of the modulation block is a major feature of the DPS-M7. This function is capable of controlling the modulation block parameter in real time with the envelope waveform created in the envelope block. (See "General Block Diagram" on page 10.) Since the parameters that can be controlled will differ according to the algorithm used, refer to the block diagram and parameter list for each algorithm.
In the block diagram for each algorithm, the sections to which this function can be applied are shown as in Fig. 1 and 2.

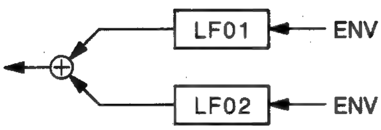


Fig. 1

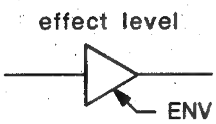


Fig. 2

Fig. 1 shows that signals from the envelope block can control LF0 (Low Frequency Oscillator) 1 and LF02, and Fig. 2 shows that signals from the envelope block can control "effect level." In the envelope block, envelope follower, envelope generator 1 (EG1) or envelope generator 2 (EG2) can be selected. (For details, refer to the envelope block on page 50).

The expression **"modulation"** such as in "frequency **modulation**" and "depth **modulation**" which indicates the effect by depth is also used as a parameter name. **"mod phase"** such as in "frequency **mod phase"** and "depth **mod phase"** expresses the phase (noninverting, inverting) of the effect depth. For example, if an envelope signal is input from the envelope block as shown in Fig. 3, the signal will be processed uninverted (Fig. 4) in the modulation block when "mod phase" is set to "normal," and will be processed inverted when "mod phase" is set to "inverse" (Fig. 5).

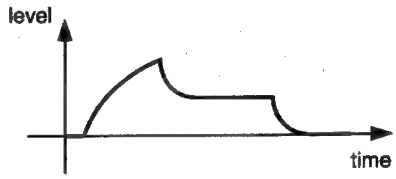


Fig. 3

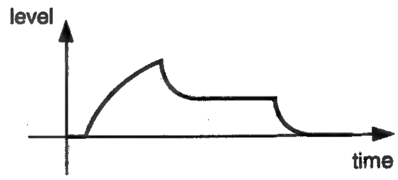


Fig. 4

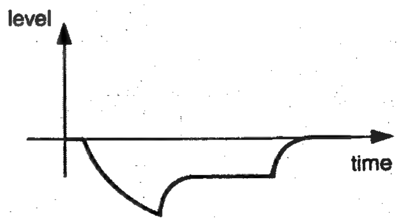


Fig. 5

To be continued ►

In summary, as shown in Fig. 6, an envelope waveform created in the envelope block is given amplitude and phase before being applied to each parameter. Amplitude is expressed by the parameter called "modulation" with a setting range of 0 - 100%. When it is set to 100%, the signal from the envelope block is applied directly to each parameter. Phase expressed by the parameter "mod phase" is also applied to each parameter.

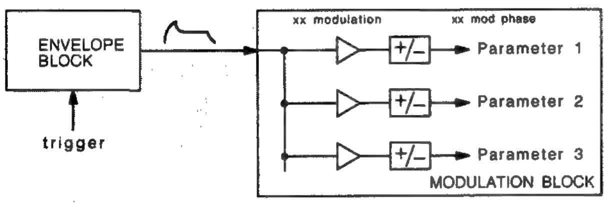


Fig. 6

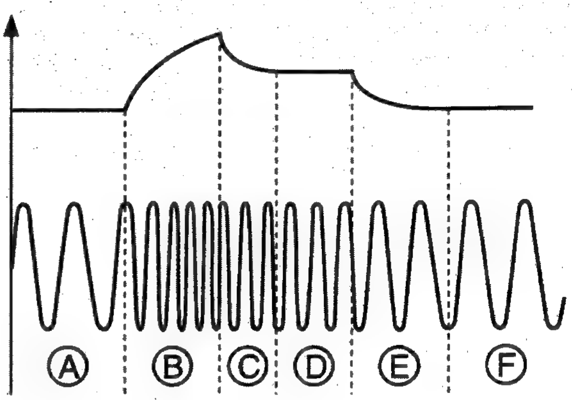
Setting Range of Fundamental Dynamic Modulation Functions

freq (frequency) modulation/freq mod (frequency modulation) phase

This parameter modulates the LFO (Low frequency Oscillator) or OSC (Oscillator) frequency. The frequency value becomes the same value set by the "frequency" parameter when "freq modulation" is set to 0%, and becomes the maximum value (double the value set by the "frequency" parameter) when "freq modulation" is set to +100%.

Moreover, the value of the frequency becomes the minimum value of 0 Hz when "freq mod phase" is set to "inverse." ("freq modulation" can be considered to be set to –100%). In other words, frequency varies between 0 to twice the value set by the "frequency" parameter according to the values set by "freq modulation" and "freq mod phase."

When envelope generator is selected in the envelope block, this feature is illustrated as in Fig. 7.



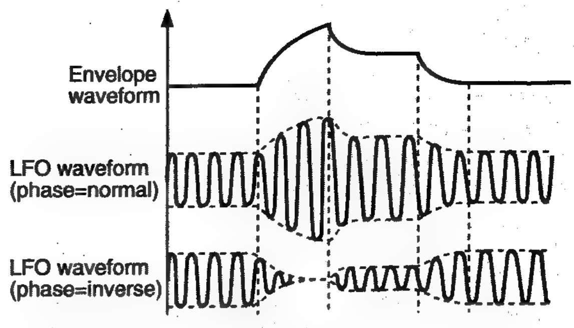
Ⓐ Normal state (frequency of set value)
Ⓑ Frequency rises gradually
Ⓒ Frequency drops to a certain point
Ⓓ Fixed frequency sustained (frequency set with "sustain level")
Ⓔ Frequency drops gradually
Ⓕ Returns to normal state (frequency of set value)

• Complicated envelope waveforms can be created by setting the parameters in the envelope block.

Fig. 7

depth modulation/depth mod (modulation) phase
This parameter modulates the effect depth.
The value of the depth becomes the same value set by the "depth" parameter when "depth modulation" is set to 0%, and becomes the maximum value (double the value set by the "depth" parameter) when "depth modulation" is set to +100%.
Moreover, the value of the depth becomes the minimum value of 0 when "depth mod phase" is set to "inverse." ("depth modulation" can be considered to be set to –100%.)
In other words, depth varies within the range of 0 to double the value set by the "depth" parameter according to the values set by "depth modulation" and "depth mod phase."

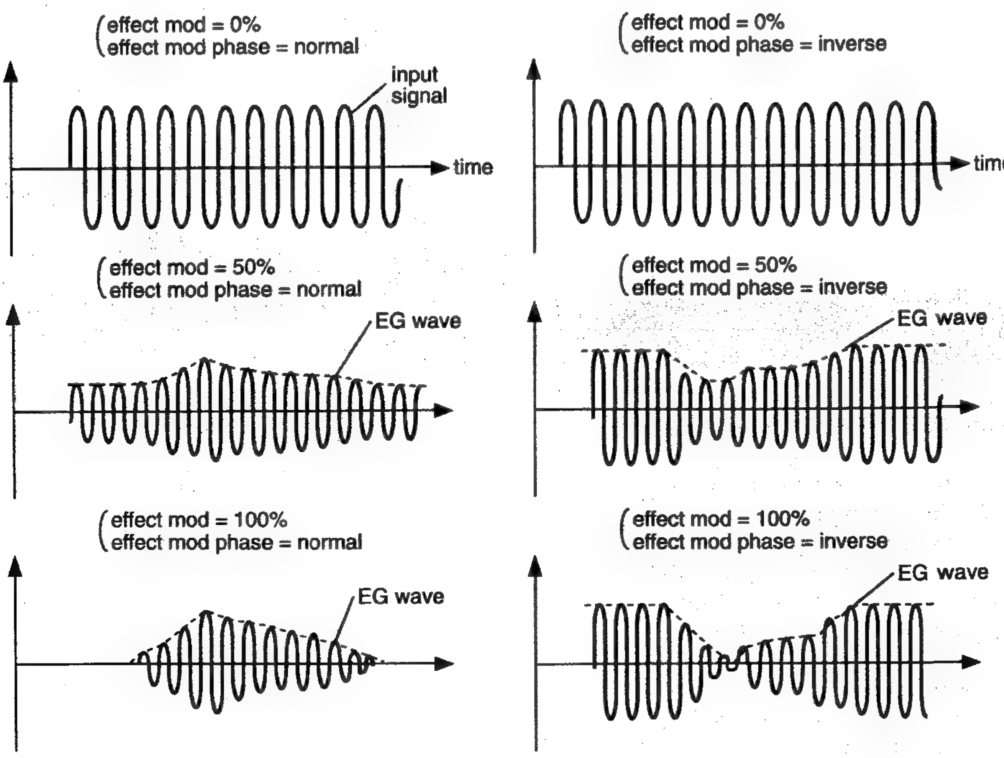
When modulating the LFO depth, this feature is illustrated as in Fig. 8 to the right.



• Complicated envelope waveforms can be created by setting the parameters in the envelope block.

Fig. 8

effect modulation/effect mod (modulation) phase
This parameter modulates the effect level.
Amplitude modulation is applied in the positive direction as the "effect modulation" level advances to the positive side ("effect mod phase" = "normal"), whereas, amplitude modulation is applied in the negative direction on the negative side ("effect mod phase" = "inverse"). When "effect modulation" level is 0%, there will be no modulation.
Unlike other dynamic modulation functions, the maximum amplitude becomes the same as that of the input signal in the case of effect modulation.



• Complicated envelope waveforms can be created by setting the parameters in the envelope block.

To be continued ►

Fundamental dynamic modulation functions are described on the previous page. For other modulation functions, refer to each algorithm.

Note:
There will be cases when the anticipated effects cannot be realized even when the "modulation" level is set to the maximum value (100%). This is because of the low amplitude of the signal from the envelope block. It is therefore preferable to receive as large a signal as possible from the envelope block to obtain the maximum effect.

Indirect Parameter (Macro Function)

The DPS-M7 has macro functions capable of simultaneous change of multiple parameters during editing by previously set rules. These are called indirect parameters and include parameters such as "sync" and "scale." (Indirect parameters differ according to the algorithm.)

The "sync" parameter is included in the parameters for which the value can be set for each channel (ch1, ch2). This parameter is indicated by "sync" added to the parameter name and is capable of changing the same values simultaneously in each channel. If different values are set in each channel, "separate" will be indicated in the set value of the sync parameter. Moreover, if "sync" is executed when different values are used in ch1 and ch2, ch2 will acquire the value of ch1.

The "scale" parameter is an indirect parameter that changes common parameters in the modulation block all at once. Simultaneous change of common parameters is possible at the indicated percentage. (The value at the time when "scale" is selected is considered to be 100% value) "over" will be displayed if scale is changed and if any one of the parameters reaches its upper limit value.

Types of scale parameters
(1) time scale
Changes the "delay time" parameters simultaneously.
(2) frequency scale
Changes "frequency" of each LFO simultaneously.
(3) depth scale
Changes "depth" (effectiveness) of each LFO simultaneously.

In addition to the above, the pitch shifter algorithms (algorithm 5 to 7) have special indirect parameters. Refer to "Pitch Shifter Macro Functions" (page 28) for these parameters.

General Parameter Description

Each algorithm has parameters corresponding to its composition.
Of the parameters that can be edited in each algorithm, the most frequently used parameters will be explained here.
There are also special parameters depending on the algorithm. Refer to "Parameter description" for these parameters.

• LFO section

LFO frequency
Sets the LFO (Low Frequency Oscillator) frequency. Effect cycle speeds up when this value increases.

LFO frequency modulation (LFO freq modulation)
Sets the effect of dynamic "LFO freq modulation". See page 19 for details.

LFO frequency modulation phase (LFO freq mod phase)
Sets the effect of the dynamic "LFO freq modulation phase". Phase is uninverted when it is set to "normal" and inverted when it is set to "inverse."

LFO depth
Sets the amplitude of the LFO. The depth of effect becomes greater as this value increases.

LFO depth modulation
Sets the effect of the dynamic "LFO depth modulation". See page 20 for details.

LFO depth modulation phase (LFO depth mod phase)
Sets the effect of the dynamic "LFO depth mod phase". Phase is uninverted when it is set to "normal" and inverted when it is set to "inverse."

LFO wave form
Selects LFO waveform. There are four types of waveforms consisting of the sin ( ), triangle ( ), special 1 ( ) and special 2 ( ) waves.

LFO phase
Sets the phase angle of the LFO. Setting range is between 0 to 359° in 1° unit. Effect with the feeling of spaciousness can also be obtained even with monaural signals by shifting the LFO phase between channels.

• Delay section

direct delay time
Sets the delay time of direct sound. The time can be set in sampling cycle units (about 21 µs).

predelay time
Sets the delay time in the preceding stage of each effect unit. The time can be set in sampling cycle units (about 21 µs).

postdelay time
Sets the delay time in the post-stage of each effect unit. The time can be set in sampling cycle units (about 21 µs).

main delay time
Sets the delay time of the main delay unit of the algorithm. The time can be set in sampling cycle units (about 21 µs).

main delay tap time/sub delay tap time
Sets the delay time of tap output from the main delay unit or subdelay unit of the algorithm. The time can be set in sampling cycle units (about 21 µs).

main delay tap level
Sets the tap level output from the main delay unit.

main delay tap phase
Sets the phase of the tap output from the main delay unit.

sub delay tap level
Sets the tap level output from the sub delay unit.

sub delay tap phase
Sets phase of the tap output from the sub delay unit.

main delay tap panpot
Sets the tap panpot output from the main delay unit. The tap panpot is localized in the channel of setting side when this parameter is set to 0%, and in the opposite channel when it is set to 100%.

sub delay tap panpot
Sets the tap panpot output from the sub delay unit. The panpot is localized in the channel of setting side when this parameter is set to 0%, and in the opposite channel when it is set to 100%.

feedback level
Sets the feedback level of each delay unit. Distortion may occur depending on the input signal or the set value for the delay time.

feedback phase
Sets the phase of feedback level. The phase is uninverted when this parameter is set to "normal" and inverted when it is set to "inverse."

high frequency feedback damping (hi freq fdbk damping)
Sets the amount of damping in the high frequency range of feedback sound. Damping amount is 0% when this parameter is set to "1.000."
The amount of damping in the high frequency range increases with the decrease of this value.

• Level section

direct level
Sets the output level of direct sound.

direct phase
Sets the phase of the output level of direct sound.

effect level
Sets the total output level of effect sound.

effect phase
Sets the phase of the total output level of effect sound.

effect modulation
Sets the effect depth of dynamic "effect modulation." (See page 20 for details)

effect modulation phase (effect mod phase)
Sets the effect level of dynamic "effect mod phase." Phase is uninverted when this parameter is set to "normal" and inverted when it is set to "inverse."

output level
Sets the output level of each effect unit.

output phase
Sets the phase of the output level of each effect unit.

panpot level
Sets the panpot output from each effect unit. The panpot is localized in the channel of setting side when this parameter is set to 0%, and in opposite channel when it is set to 100%.

• EQ section
Refer to page 13 as for the equalizer parameters.

To be continued ►

Algorithm 0 Effect Off OFF

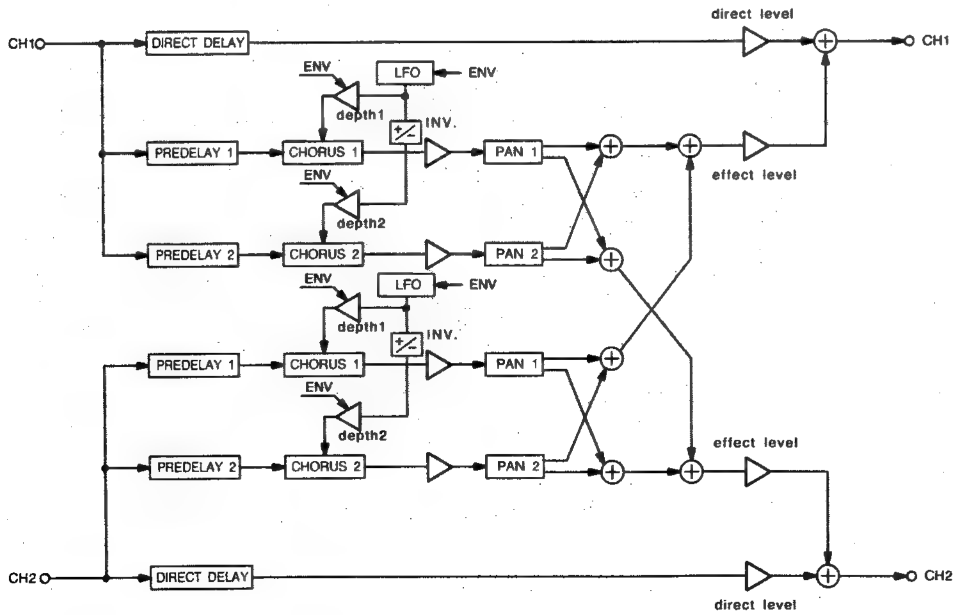
The modulation block will be passed though so that no effect can be obtained in this block.

Algorithm 1 Stereo Chorus SCH

This algorithm produces a stereo chorus effect in which the high frequency range has been accentuated by over-sampling processing (sampling rate: 96 kHz) used in this algorithm. There are two chorus blocks in each channel. The uninverted output of LFO is input to the Chorus 1 unit and the inverted output is input to the Chorus 2 unit. Depth (effectiveness) can be set in each unit. Chorus effect with the feeling of spaciousness can be achieved even with monaural signals by shifting the LFO phases between two channels.

Parameter description
LFO frequency
Sets the frequency of the LFO.
LFO depth
Sets the effect depth of the LFO. This parameter modulates the delay time for approximately 20 msec when it is set to 100%.
LFO wave form
Selects the waveform of the LFO. There are four types of waveforms consisting of sin (⌒◡), triangle (/\/), special 1(⌒⌒) and special 2 (◡◡).
panpot
Localized in same channel as the signal input when set to 0% and in the opposite channel when set to 100%.

Section	Parameter	MIN and MAX
	chorus on/off	on/off
LFO section	LFO frequency (ch1, ch2)	0.01 – 40.00Hz
	LFO freq modulation (ch1, ch2)	0 – 100.00%
	LFO freq mod phase (ch1, ch2)	normal/inverse
	LFO depth 1, 2 (ch1, ch2)	0 – 100.00%
	LFO depth 1, 2 modulation (ch1, ch2)	0 – 100.00%
	LFO depth 1, 2 mod phase (ch1, ch2)	normal/inverse
	LFO wave form (ch1, ch2)	sin/triangle/special1/special2
	LFO phase (ch1, ch2)	0 – 359°
DELAY section	direct delay time (ch1, ch2)	0 – 300.00msec
	predelay 1, 2 time (ch1, ch2)	0 – 300.00msec
LEVEL section	output 1, 2 level (ch1, ch2)	0 – 100.00%
	output 1, 2 phase (ch1, ch2)	normal /inverse
	panpot 1, 2 level (ch1, ch2)	0 – 100.00%
	direct level (ch1, ch2)	0 – 100.00%
	direct phase (ch1, ch2)	normal/inverse
	effect level (ch1, ch2)	0 – 100.00%
	effect phase (ch1, ch2)	normal/inverse

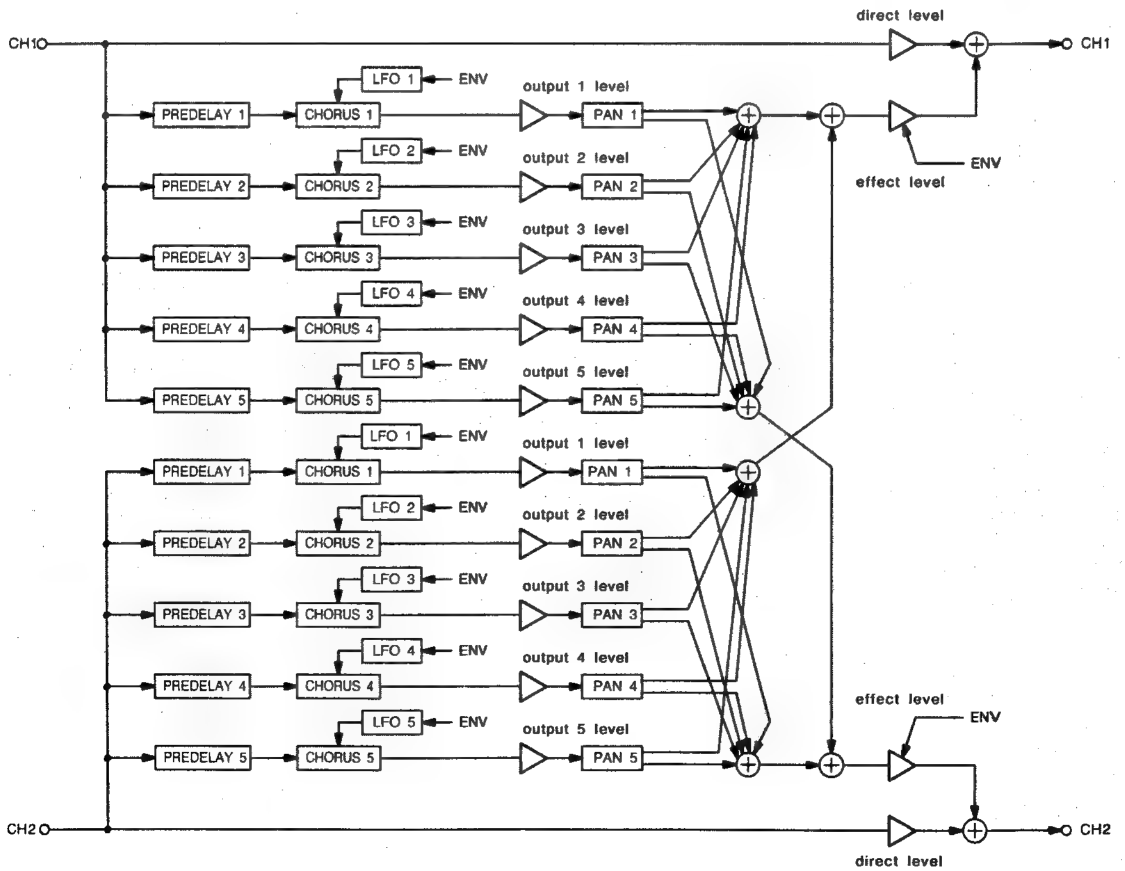


Algorithm 2 Deca Chorus DCH

This is a chorus program. Since each channel has 5 chorus units, variegated sounds can be produced such as 5-phase stereo chorus during stereo inputs and 10-phase monaural chorus during monaural inputs. Although "LFO frequency" is set to the same value for all ten LFO's, "LFO depth" is set independently by channel and "LFO phase" can be set for each LFO. Therefore, sounds with the feeling of spaciousness can be achieved even with monaural sources.

Section	Parameter	MIN and MAX
	chorus on/off	on/off
LFO section	LFO frequency LFO freq modulation LFO freq modulation phase	0.01 – 40.00Hz 0 – 100.00% normal/inverse
	LFO depth (ch1, ch2) LFO depth modulation (ch1,ch2) LFO depth mod phase (ch1,ch2)	0 – 100.00% 0 – 100.00% normal/inverse
	LFO wave form (ch1, ch2) LFO 1 – 5 phase (ch1, ch2)	sin/triangle/special1/special2 0 – 359*
DELAY section	predelay 1 – 5 time (ch1, ch2)	0 – 1000.00msec
LEVEL section	output 1 – 5 level (ch1, ch2) output 1 – 5 phase (ch1, ch2) panpot 1 – 5 level (ch1, ch2)	0 – 100.00% normal /inverse 0 – 100.00%
	direct level (ch1, ch2) direct phase (ch1, ch2) effect level (ch1, ch2) effect phase (ch1, ch2) effect modulation (ch1, ch2) effect mod phase (ch1, ch2)	0 – 100.00% normal/inverse 0 – 100.00% normal/inverse 0 – 100.00% normal/inverse

* The delay time for approximately 20 msec when "LFO depth" is set to 100%.



To be continued ►

Algorithm 3 Multi Chorus MCH

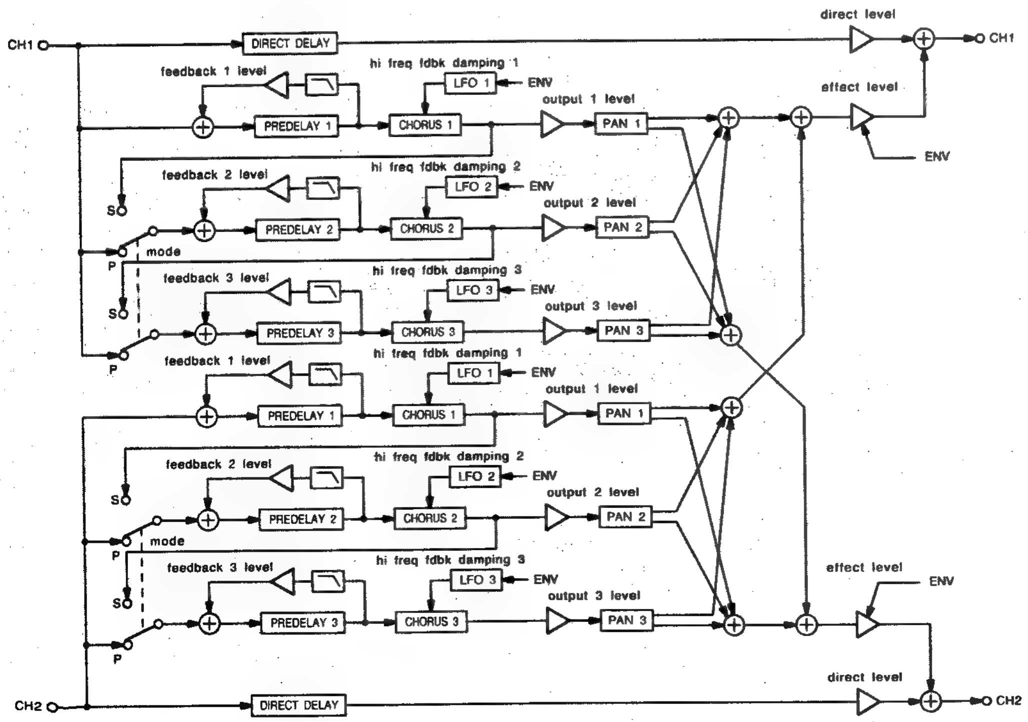
This is a chorus program. Each channel has three chorus units with feedback delays.
Serial or parallel connection of these three units can be selected with "chorus mode". Damping of the high frequency range is also possible with "hi freq fdbk damping" in the feedback loop.

Parameter description

high frequency feedback damping (hi freq fdbk damping)
Sets the damping response in the high frequency range. When it is set to 1.000, the damping response remains unchanged and the amount of damping increases as this value decreases.

LFO depth
Sets the effect depth of the LFO. This parameter modulates the delay time for approximately 20 msec when it is set to 100%.

Section	Parmeter	MIN and MAX
	chorus on/off	on/off
	chorus mode	parallel/serial
LFO section	LFO 1 – 3 frequency (ch1, ch2) LFO 1 – 3 freq modulation(ch1,ch2) LFO 1 – 3 freq mod phase(ch1,ch2)	0.01 – 40.00Hz 0 – 100.00% normal/inverse
	LFO 1 – 3 depth (ch1, ch2) LFO 1 – 3 depth modulation (ch1, ch2) LFO 1 – 3 depth mod phase (ch1, ch2)	0 – 100.00% 0 – 100.00% normal/inverse
	LFO 1 – 3 wave form (ch1, ch2) LFO 1 – 3 phase (ch1, ch2)	sin/triangle/ special1/special2 0 – 359°
DELAY section	direct delay time (ch1, ch2) predelay 1 – 3 time (ch1, ch2)	0 – 300.00msec 0 – 300.00msec
	feedback 1 – 3 level (ch1, ch2) feedback 1 – 3 phase (ch1, ch2) hi freq fdbk damping 1-3 (ch1,ch2)	0 – 99.90% normal/inverse 0.003 – 1.000
LEVEL section	output 1 – 3 level (ch1, ch2) output 1 – 3 phase (ch1, ch2) panpot 1 – 3 level (ch1, ch2)	0 – 100.00% normal/inverse 0 – 100.00%
	direct level (ch1, ch2) direct phase (ch1, ch2) effect level (ch1, ch2) effect phase (ch1, ch2) effect modulation (ch1, ch2) effect mod phase (ch1, ch2)	0 – 100.00% normal/inverse 0 – 100.00% normal/inverse 0 – 100.00% normal/inverse



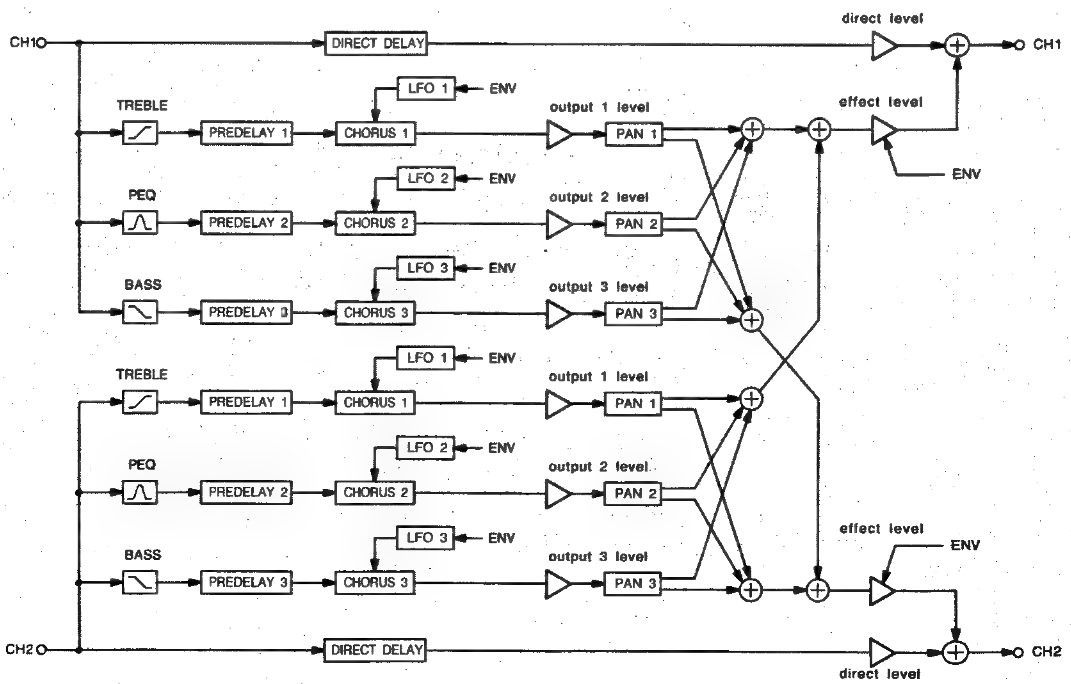
Algorithm 4 Band Chorus BCH

This is a chorus program in which the input signal is divided into three signals. Each of these divided signals is then sent to an EQ unit and a chorus unit. Independent settings in each unit enables you to create your own desired sounds.

Section	Parameter	MIN and MAX
	chorus on/off	on/off
EQ section	treble frequency (ch1, ch2) treble level (ch1, ch2) PEQ frequency (ch1, ch2) PEQ level (ch1, ch2) PEQ q (ch1, ch2) bass frequency (ch1, ch2) bass level (ch1, ch2)	400Hz–20.0kHz –12 – +12dB 63Hz – 20.0kHz –12 – +12dB 0.267 – 34.62 16Hz – 6.3kHz –12 – +12dB
LFO section	LFO 1 – 3 frequency (ch1, ch2) LFO 1 – 3 freq modulation (ch1, ch2) LFO 1 – 3 freq mod phase (ch1, ch2)	0.01 – 40.00Hz 0 – 100.00% normal/inverse
	LFO1 – 3 depth (ch1, ch2) LFO 1–3 depth modulation(ch1,ch2) LFO 1 – 3 depth mod phase (ch1, ch2)	0 – 100.00% 0 – 100.00% normal/inverse
	LFO 1 – 3 wave form (ch1, ch2) LFO 1 – 3 phase (ch1, ch2)	sin/triangle/ special1/special2 0 – 359˚

Section	Parameter	MIN and MAX
DELAY section	direct delay time (ch1, ch2) predelay 1 – 3 time (ch1, ch2)	0 – 300.00msec 0 – 300.00msec
LEVEL section	output 1 – 3 level (ch1, ch2) output 1 – 3 phase (ch1, ch2) panpot 1 – 3 level (ch1, ch2)	0 – 100.00% normal/inverse 0 – 100.00%
	direct level (ch1, ch2) direct phase (ch1, ch2) effect level (ch1, ch2) effect phase (ch1, ch2) effect modulation (ch1, ch2) effect mod phase (ch1, ch2)	0 – 100.00% normal/inverse 0 – 100.00% normal/inverse 0 – 100.00% normal/inverse

* The delay time for approximately 20 msec when "LFO depth" is set to 100%.



To be continued ►

Concerning Pitch Shifter

The amount of pitch shift can be controlled by the MIDI note data in pitch shifter algorithms 5, 6 and 7. Furthermore, since the amount of shift can be set according to the input note (C, C#, D, ... , B), it is possible to create pitch shift effects matching the scale of the tune. There are two pitch shifter modes.

(1) Fixed mode
Shifts the pitch of the input signal with a fixed amount (set value).

(2) Harmony mode
Capable of changing the amount of shift in compliance with 12 types of notes (C, C#, D, ... , B) input by the MIDI note data. For example, if a keyboard and the DPS-M7 are connected as shown in Fig. 1 and if the parameters are set as in Table 1, the DPS-M7 outputs pitch shift sounds of three degrees above the actually input sounds in the C-major scale. (Fig.2, Fig. 3)

By setting the amount of shift relative to each note in the "note interval" parameter, pitch shift matching of the scale of the tune or matching a special scale is possible.

Parameter	Meaning
note interval C	+400 cent
note interval C #	0 cent
note interval D	+300 cent
note interval D #	0 cent
note interval E	+300 cent
note interval F	+400 cent
note interval F #	0 cent
note interval G	+400 cent
note interval G #	0 cent
note interval A	+300 cent
note interval A #	0 cent
note interval B	+300 cent

Table 1

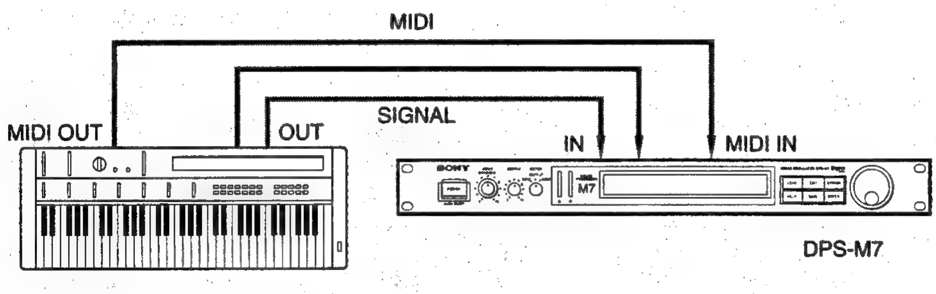


Fig.1 Connection

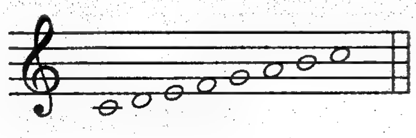


Fig.2 Sound from the keyboard

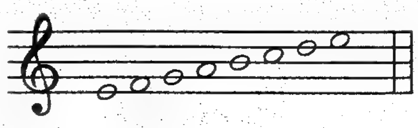


Fig.3 Pitch shifted sound

Pitch Shifter Macro Functions

Although a "note interval" can be set for each note in harmony mode, the pitch shifter has exclusive macro functions (indirect parameters) to save the time of resetting according to the key, scale, and interval (the amount of shift) of the tune being played. Three indirect parameters are listed below.

(1) scale type
Sets the scale of the tune. 7 types of scales are initially prepared. They consist of Major, Dorian, Phrygian, Lydian, Mixo-Lydian, Aeolian = Natural Minor and Locrian. When setting to a scale other than these, select "user" in this parameter.

(2) base key
Sets the key of the tune. 12 types of C, C#, D ... A#, B are initially prepared.

Note:
"base key" cannot be set if "user" is selected in "scale type."

(3) note interval
Sets amount of shift by assigning the number of degrees above the scale. In other words, indicate the amount of shift with the degrees above (or below) the scale.

Note:
"note interval" cannot be set if "user" is selected in "scale type." As explained, the amount of shift (cent) of direct parameters "note interval C" to "note interval B" are automatically set by indirect parameters (1) to (3) above.

Examples of how direct parameters are actually set with the macro functions
Example 1) When "base key" is set to C and "note interval"= +3rd.

• Notes out of the scale range are set to 0 cent.

Note	Scale type						
Interval	Major	Dorian	Phrygian	Lydian	Mixo-Lydian	Aeolian	Locrian
C	+400	+300	+300	+400	+400	+300	+300
C #	0	0	+400	0	0	0	+400
D	+300	+300	0	+400	+300	+400	0
D #	0	+400	+400	0	0	+400	+300
E	+300	0	0	+300	+300	0	0
F	+400	+400	+300	0	+400	+300	+300
F #	0	0	0	+300	0	0	+400
G	+400	+300	+300	+400	+300	+300	0
G #	0	0	+400	0	0	+400	+400
A	+300	+300	0	+300	+300	0	0
A #	0	+400	+300	0	+400	+400	+300
B	+300	0	0	+300	0	0	0

Example 2) When "base key" is set to A and "note interval" = –4th.

Note	Scale type						
Interval	Major	Dorian	Phrygian	Lydian	Mixo-Lydian	Aeolian	Locrian
C	0	–500	–500	0	0	–500	–500
C #	–500	0	0	–500	–600	0	0
D	–500	–500	–500	0	–500	–500	–500
D #	0	0	0	–600	0	0	–500
E	–500	–500	–600	–500	–500	–500	0
F	0	0	–500	0	0	–500	–500
F #	–500	–600	0	–500	–500	0	0
G	0	–500	–500	0	–500	–500	–500
G #	–600	0	0	–500	0	0	0
A	–500	–500	–500	–500	–500	–500	–600
A #	0	0	–500	0	0	0	–500
B	–500	–500	0	–500	–500	–600	0

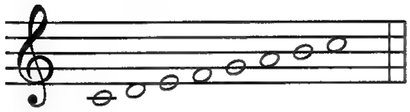
• Notes out of the scale range are set to 0 cent.

To be continued ▶

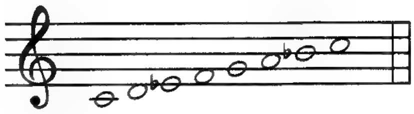
Concerning the Scale

The seven scales selectable with “scale type” are indicated below (when “base key” is set to C).

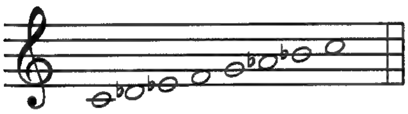
(1) Major scale



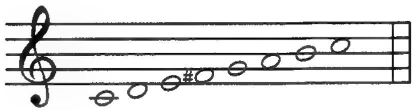
(2) Dorian scale



(3) Phrygian scale

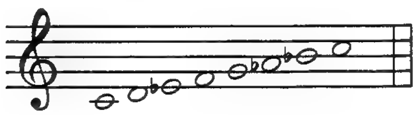


(4) Lydian scale

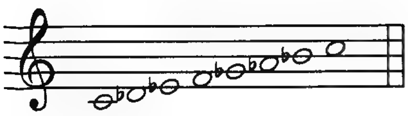


(5) Mixo-Lydian scale

(6) Aeolian scale (=Natural Minor scale)



(7) Locrian scale



Algorithm 5 Stereo Pitch Shifter SPS

This is an algorithm in which both ch1 and ch2 have a high quality pitch shifter unit. Since a delay unit that can be set to a maximum of 1,000 msec is also provided in each channel, a variety of sounds can be created in combination with the feedback delay.

Parameter description

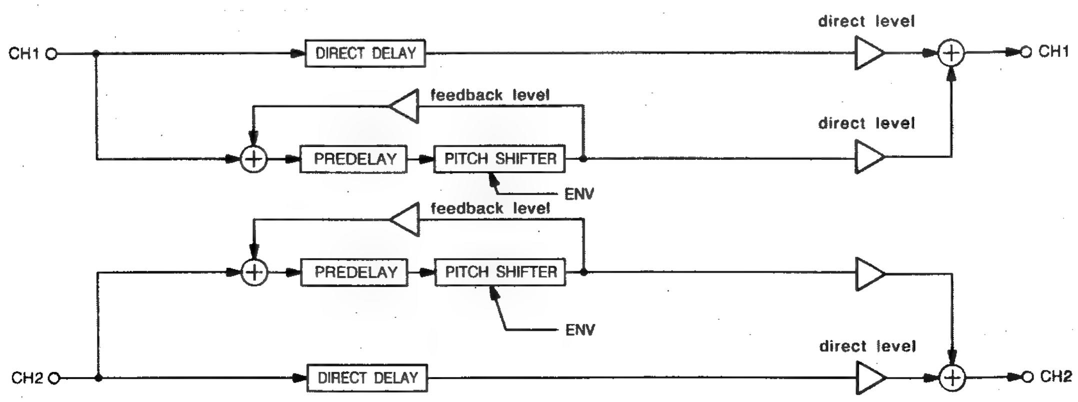
pitch modulation

Changes the amount of pitch shift. The amount of pitch shift becomes the same value set by "pitch" in fixed mode (or "note interval" in harmony mode) when this parameter is set to 0%. When this parameter is set to +100% with "pitch" (or "note interval") set to 0 cent, the amount of shift becomes its maximum + one octave. Conversely, when it is set to –100% in the same condition, the amount of pitch shift decreases down to – ∞(infinity) just as if the input signal were stopped.

Section	Parameter	MIN and MAX
	pitch shifter on/off	on/off
	pitch shifter mode	fixed/harmony
PITCH section (fixed)	pitch (ch1, ch2)	–2400 – +2400cent
PITCH section (harmony)	note interval C (ch1, ch2)	–2400 – +2400cent
	note interval C# (ch1, ch2)	–2400 – +2400cent
	note interval D (ch1, ch2)	–2400 – +2400cent
	note interval D# (ch1, ch2)	–2400 – +2400cent
	note interval E (ch1, ch2)	–2400 – +2400cent
	note interval F (ch1, ch2)	–2400 – +2400cent
	note interval F# (ch1, ch2)	–2400 – +2400cent
	note interval G (ch1, ch2)	–2400 – +2400cent
	note interval G# (ch1, ch2)	–2400 – +2400cent
	note interval A (ch1, ch2)	–2400 – +2400cent
	note interval A# (ch1, ch2)	–2400 – +2400cent
	note interval B (ch1, ch2)	–2400 – +2400cent
PITCH section (common)	pitch modulation (ch1, ch2)	0 – 100.00%
	pitch mod phase (ch1, ch2)	normal/inverse
DELAY section	direct delay time (ch1, ch2)	0 – 250.00msec
	predelay time (ch1, ch2)	0 – 1000.00msec
	feedback level (ch1, ch2)	0 – 99.90%
	feedback phase (ch1, ch2)	normal/inverse
LEVEL section	direct level (ch1, ch2)	0 – 100.00%
	direct phase (ch1, ch2)	normal/inverse
	effect level (ch1, ch2)	0 – 100.00%
	effect phase (ch1, ch2)	normal/inverse

Note

If "pitch" (or "note interval") is set to "sync," the values set for ch2 in "pitch modulation" and "pitch mod phase" will be ignored and the values set for ch1 will also become valid for ch2.



To be continued ▶

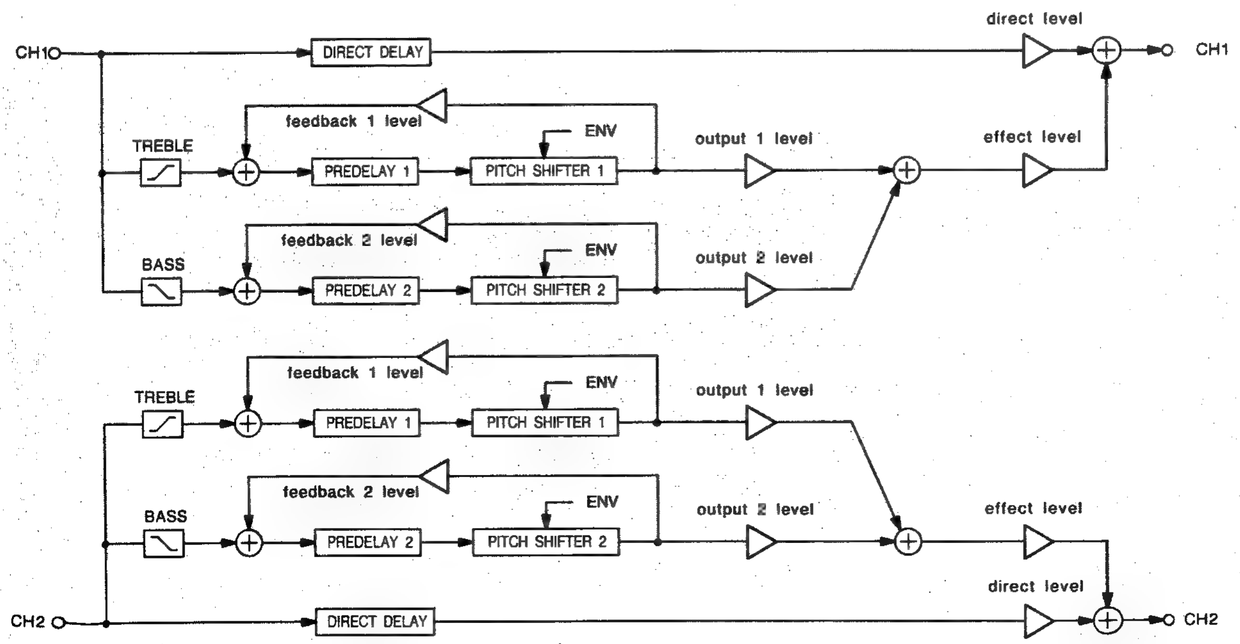
Algorithm 6 Band Pitch Shifter BPS

This is a pitch shifter program in which the input signal is divided into two signals.
Each of these divided signals is then sent to an EQ unit and a pitch shifter unit.

Section	Parameter	MIN and MAX
	pitch shifter on/off	on/off
	pitch shifter mode	fixed/harmony
EQ section	treble frequency (ch1, ch2) treble level (ch1, ch2) bass frequency (ch1, ch2) bass level (ch1, ch2)	400Hz – 20.0kHz –12 – +12dB 16Hz – 6.3kHz –12 – +12dB
PITCH section (fixed)	pitch 1, 2 (ch1, ch2)	–2400 – +2400 cent
PITCH section (harmony)	note interval 1, 2 C (ch1, ch2) note interval 1, 2 C # (ch1, ch2) note interval 1, 2 D (ch1, ch2) note interval 1, 2 D # (ch1, ch2) note interval 1, 2 E (ch1, ch2) note interval 1, 2 F (ch1, ch2) note interval 1, 2 F # (ch1, ch2) note interval 1, 2 G (ch1, ch2) note interval 1, 2 G # (ch1, ch2) note interval 1, 2 A (ch1, ch2) note interval 1, 2 A # (ch1, ch2) note interval 1, 2 B (ch1, ch2)	–2400 – +2400cent –2400 – +2400cent –2400 – +2400cent –2400 – +2400cent –2400 – +2400cent –2400 – +2400cent –2400 – +2400cent –2400 – +2400cent –2400 – +2400cent –2400 – +2400cent –2400 – +2400cent –2400 – +2400cent

Section	Parameter	MIN and MAX
PITCH section (common)	pitch modulation 1, 2(ch1, ch2) pitch mod 1, 2 phase(ch1, ch2)	0 – 100.00% normal/inverse
DELAY section	direct delay time (ch1, ch2) predelay 1, 2 time (ch1, ch2)	0 – 180.00msec 0 – 500.00msec
	feedback 1, 2 level (ch1, ch2) feedback 1, 2 phase (ch1, ch2)	0 – 99.90% normal/inverse
LEVEL section	output 1, 2 level (ch1, ch2) output 1, 2 phase (ch1, ch2)	0 – 100.00% normal/inverse
	direct level (ch1, ch2) direct phase (ch1, ch2) effect level (ch1, ch2) effect phase (ch1, ch2)	0 – 100.00% normal/inverse 0 – 100.00% normal/inverse

Note
If "pitch" (or "note interval") is set to "sync," the values set for ch2 in "pitch modulation" and "pitch mod phase" will be ignored and the values set for ch1 will also become valid for ch2.



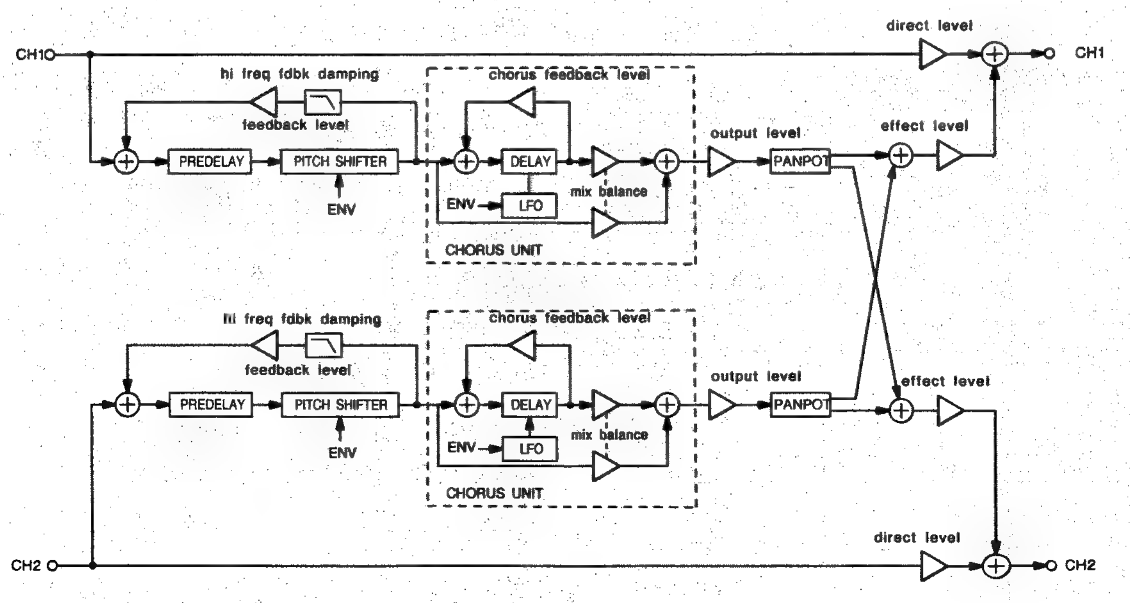
Algorithm 7 Pitch Shift Modulation PSM

This is a pitch shifter algorithm. A chorus unit with feedback is located in each channel at the post-stage of the pitch shifter unit. Voluminous sounds can be created by adding a chorus effect to the pitch shifted sound.

Section	Parameter	MIN and MAX
	pitch shift mod on/off	on/off
	pitch shifter mode	fixed/harmony
PITCH section (fixed)	pitch (ch1, ch2)	–2400 – +2400cent
PITCH section (harmony)	note interval C (ch1, ch2)	–2400 – +2400cent
	note interval C# (ch1, ch2)	–2400 – +2400cent
	note interval D (ch1, ch2)	–2400 – +2400cent
	note interval D# (ch1, ch2)	–2400 – +2400cent
	note interval E (ch1, ch2)	–2400 – +2400cent
	note interval F (ch1, ch2)	–2400 – +2400cent
	note interval F# (ch1, ch2)	–2400 – +2400cent
	note interval G (ch1, ch2)	–2400 – +2400cent
	note interval G# (ch1, ch2)	–2400 – +2400cent
	note interval A (ch1, ch2)	–2400 – +2400cent
	note interval A# (ch1, ch2)	–2400 – +2400cent
	note interval B (ch1, ch2)	–2400 – +2400cent
PITCH section (common)	pitch modulation (ch1, ch2)	0 – 100.00%
	pitch mod phase (ch1, ch2)	normal/inverse
DELAY section	predelay time (ch1, ch2)	0 – 1000.00msec
	feedback level (ch1, ch2)	0 – 99.90%
	feedback phase (ch1, ch2)	normal/inverse
	hi freq fdbk damping (ch1, ch2)	0.003 – 1.000

Section	Parameter	MIN and MAX
CHORUS section	LFO frequency (ch1, ch2)	0.01 – 40.0Hz
	LFO freq modulation (ch1, ch2)	0 – 100.00%
	LFO freq mod phase (ch1, ch2)	normal/inverse
	LFO depth (ch1, ch2)	0 – 100.00%
	LFO depth modulation (ch1, ch2)	0 – 100.00%
	LFO depth mod phase (ch1, ch2)	normal/inverse
	LFO wave form (ch1, ch2)	sin/triangle/ special1/ special2
	LFO phase (ch1, ch2)	0 – 359°
	chorus delay time (ch1, ch2)	0 – 200.00msec
	chorus feedback level (ch1, ch2)	0 – 99.90%
	chorus feedback phase (ch1, ch2)	normal/inverse
	DRY:EFF mix balance	100:0 – 0:100
LEVEL section	output level (ch1, ch2)	0 – 100.00%
	output phase (ch1, ch2)	normal/inverse
	panpot level (ch1, ch2)	0 – 100.00%
	direct level (ch1, ch2)	0 – 100.00%
	direct phase (ch1, ch2)	normal/inverse
	effect level (ch1, ch2)	0 – 100.00%
	effect phase (ch1, ch2)	normal/inverse

Note
If "pitch" (or "note interval") is set to "sync," the values set for ch2 in "pitch modulation" and "pitch mod phase" will be ignored and the values set for ch1 will also become valid for ch2.



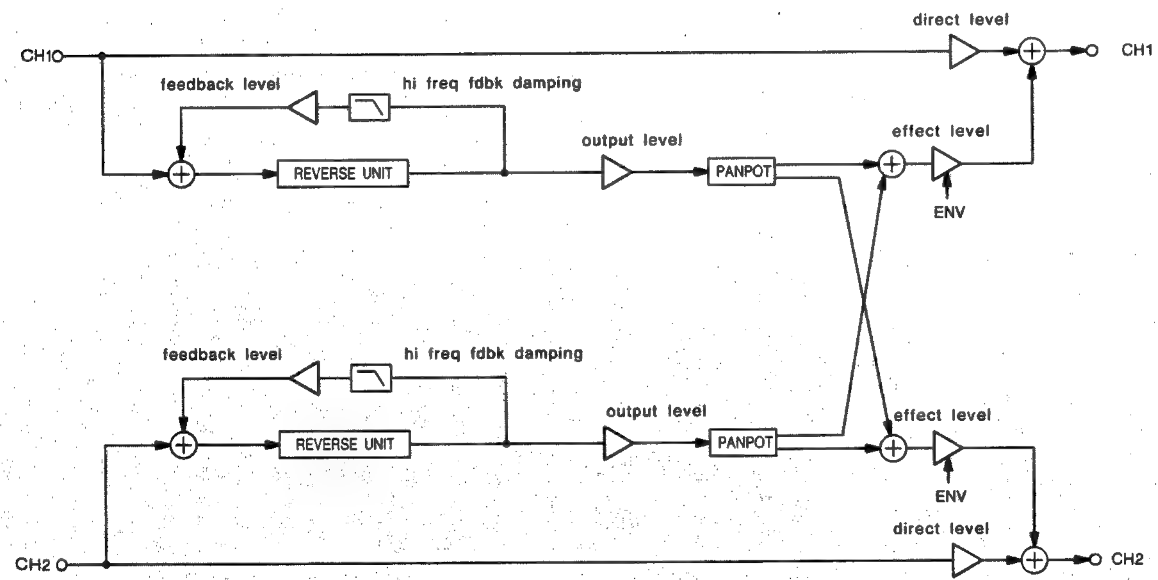
To be continued ▶

Algorithm 8 Reverse Shift RVS

This is a reverse shift algorithm in which reverse sound can be obtained. Pitch shift is possible within the range of ±1 octave.

Parameter description
pitch
Sets the amount of shift of reverse sound.
length
Sets the loop time of reverse sound.

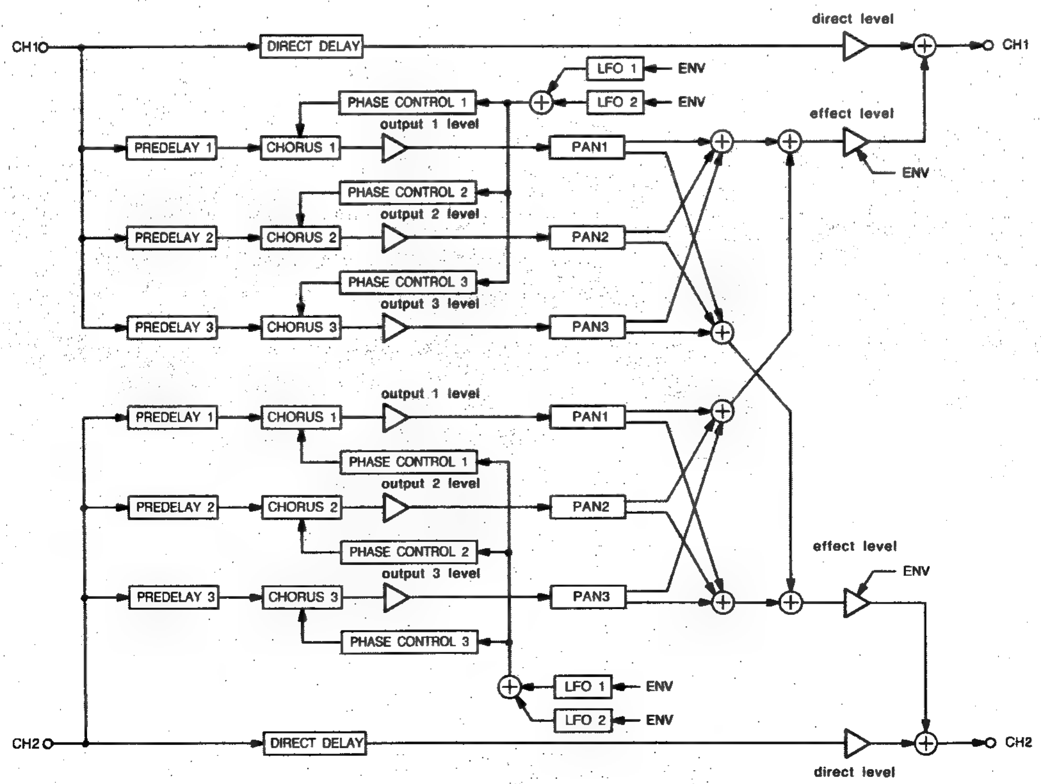
Section	Parameter	MIN and MAX
	reverse shift on/off	on/off
REVERSE section	pitch (ch1, ch2) length (ch1, ch2)	–1200 – +1200cent 10.00 –1350.00msec
	feedback level (ch1, ch2) feedback phase (ch1, ch2) hi freq fdbk damping (ch1, ch2)	0 – 99.90% normal/inverse 0.003 – 1.000
LEVEL section	output level (ch1, ch2) output phase (ch1, ch2) panpot level (ch1, ch2)	0 – 100.00% normal /inverse 0 – 100.00%
	direct level (ch1, ch2) direct phase (ch1, ch2) effect level (ch1, ch2) effect phase (ch1, ch2) effect modulation (ch1, ch2) effect mod phase (ch1, ch2)	0 – 100.00% normal/inverse 0 – 100.00% normal/inverse 0 – 100.00% normal/inverse



Algorithm 9 Stereo Ensemble ENS

This is an ensemble program in which a more voluminous effect can be obtained compared to conventional choruses by mixing two LFOs.

Section	Parameter	MIN and MAX
	ensemble on/off	on/off
LFO section	LFO 1, 2 frequency (ch1, ch2)	0.01 – 40.00Hz
	LFO 1, 2 freq modulation (ch1, ch2)	0 – 100.00%
	LFO 1, 2 freq mod phase (ch1, ch2)	normal/inverse
	LFO 1, 2 depth (ch1, ch2)	0 – 100.00%
	LFO 1, 2 depth modulation (ch1,ch2)	0 – 100.00%
	LFO 1, 2 depth mod phase (ch1,ch2)	normal/inverse
	LFO 1, 2 wave form (ch1, ch2)	sin/triangle/ special1/special2
	LFO1 phase 1 – 3 (ch1, ch2)	0 – 359°
	LFO2 phase 1 – 3 (ch1, ch2)	0 – 359°
DELAY section	direct delay time (ch1, ch2)	0 – 1000.00msec
	predelay 1 – 3 time (ch1, ch2)	0 – 1000.00msec
	output 1 – 3 level (ch1, ch2)	0 – 100.00%
	output 1 – 3 phase (ch1, ch2)	normal /inverse
	panpot 1 – 3 level (ch1, ch2)	0 – 100.00%
LEVEL section	direct level (ch1, ch2)	0 – 100.00%
	direct phase (ch1, ch2)	normal/inverse
	effect level (ch1, ch2)	0 – 100.00%
	effect phase (ch1, ch2)	normal/inverse
	effect modulation (ch1, ch2)	0 – 100.00%
	effect mod phase (ch1, ch2)	normal/inverse



To be continued ▶

Algorithm 10 Multi Phaser MPH

This is a phase shifter program with two phase shifter units in each channel. Each phase shifter can consist of a maximum of eight stages. Since the number of phase shifter stage in each unit can be freely set and the connection of the 2 phase shifters can be changed between parallel and serial with "mode," phase shifters of from one to a maximum of 16 stages can be configured. It is also possible to create new sounds since the LFO can be changed in a stair-like form with the S/H (Sample & Hold) function of the LFO waveform.

Parameter description

stage number
Sets the number of stages in the phase shifter. Phase will be delayed by a maximum of 180 ° at one stage.

mode
Sets the connection of 2 phase shifters. A parallel or serial connection can be selected.

manual
Adjusts the main phase effect frequency. Adjustments in the high frequency range are possible by increasing this value.

manual modulation
Sets the effect depth of the dynamic "manual modulation."

manual modulation phase (manual mod phase)
Sets the effect of the dynamic "manual mod phase."
Phase is uninverted when it is set to "normal" and inverted when it is set to "inverse."

S/H step frequency
Makes the LFO change in a stair-like form. A normal LFO waveform will be obtained when it is set to "off." If a frequency is set, the amount of LFO change will be held at the set frequency.

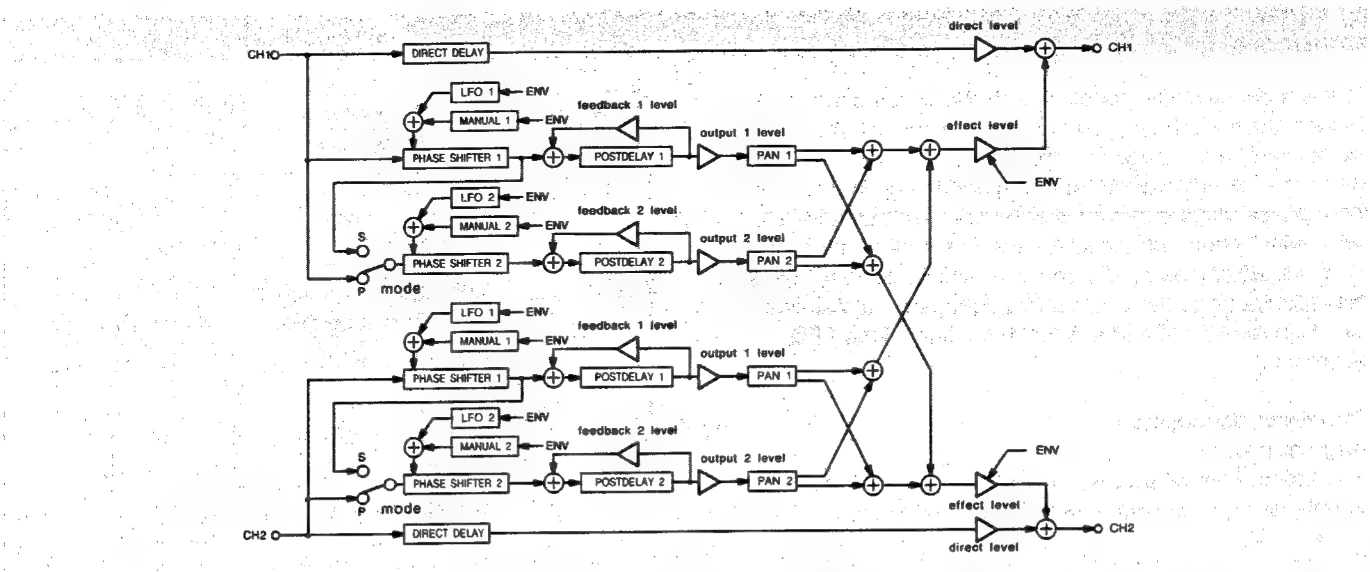
S/H step frequency modulation (S/H step freq mod)
Sets the effect depth of the dynamic "S/H step freq mod."

S/H step frequency modulation phase (S/H step f. mod phase)
Sets the effect of the dynamic "S/H step f. mod phase".
Phase is uninverted when it is set to "normal" and inverted when it is set to "inverse."

resonance level
Sets the phaser resonance level. Sounds become more distinctive as this value increases.

resonance phase
Sets the resonance phase. Phase is uninverted when it is set to "normal" and inverted when it is set to "inverse."

Section	Parameter	MIN and MAX
	phase shifter on/off	on/off
	stage number 1, 2 (ch1, ch2) mode (ch1, ch2)	pass/1/2/3/4/5/6/7/8 parallel/serial
LFO section	LFO 1, 2 frequency (ch1, ch2) LFO 1, 2 freq modulation (ch1, ch2) LFO 1, 2 freq mod phase (ch1, ch2)	0.01 – 40.00Hz 0 – 100.00% normal/inverse
	LFO 1, 2 depth (ch1, ch2)	0 – 100.00%
	manual 1, 2 (ch1, ch2) manual 1, 2 modulation (ch1, ch2) manual 1, 2 mod phase (ch1, ch2)	0 – 100.00% 0 – 100.00% normal/inverse
	S/H step frequency 1, 2 (ch1, ch2) S/H step freq mod 1, 2 (ch1, ch2) S/H step f. mod 1, 2 phase (ch1, ch2)	off/0.01–100.00Hz 0 – 100.00% normal/inverse
	LFO 1, 2 wave form (ch1, ch2) LFO1, 2 phase (ch1, ch2)	sin/triangle/special1/special2 0 – 359°
PHASE SHIFTER section	resonance 1, 2 level (ch1, ch2) resonance 1, 2 phase (ch1, ch2)	0 – 99.90% normal/inverse
DELAY section	direct delay time (ch1, ch2) postdelay 1, 2 time (ch1, ch2) feedback 1, 2 level (ch1, ch2) feedback 1, 2 phase (ch1, ch2)	0 – 350.00msec 0 – 500.00msec 0 – 99.90% normal/inverse
	output 1, 2 level (ch1, ch2) output 1, 2 phase (ch1, ch2) panpot 1, 2 level (ch1, ch2)	0 – 100.00% normal /inverse 0 – 100.00%
LEVEL section	direct level (ch1, ch2) direct phase (ch1, ch2) effect level (ch1, ch2) effect phase (ch1, ch2) effect modulation (ch1, ch2) effect mod phase (ch1, ch2)	0 – 100.00% normal/inverse 0 – 100.00% normal/inverse 0 – 100.00% normal/inverse



Algorithm 11 Stereo Flanger SFL

This algorithm produces a stereo flanger effect in which the high frequency range has been accentuated by over-sampling processing (sampling rate: 96 kHz) used in this algorithm.
Each channel is composed of a single flanger unit. Although this is extremely simple, it is a new flanger which takes into consideration setting of two LFO's, of feedback mode and of effect mode independently in each channel, and simulation of tape flangers such as 2-stage fixed phase shifters connected to the direct lines.

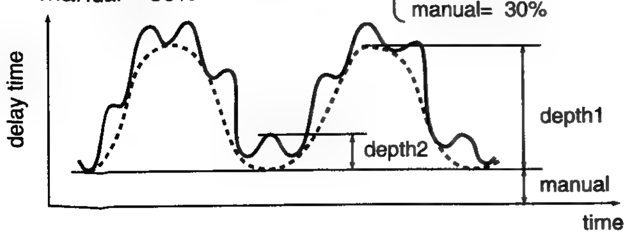
Parameter description

LFO depth
Sets the effect depth of the LFO.
Delay time is approximately 20 msec for one LFO when "depth" is set to 100%.

manual
Sets the main frequency of effect. Delay time increases as the value for "manual" becomes larger. A maximum delay time of approximately 63 msec can be obtained when "LFO 1 depth," "LFO 2 depth" and "manual" are all set to the maximum.

Example:
A maximum delay time of approximately 26.25 msec will be obtained with the following settings.
LFO 1 depth = 100%
LFO 2 depth = 25%
manual = 30%

depth1=100% (LFO1)
depth2= 25% (LFO2)
manual= 30%



manual modulation
Sets the effect depth of the dynamic "manual modulation."

manual modulation phase (manual mod phase)
Sets the effect of the dynamic "manual mod phase." Phase is uninverted when it is set to "normal" and inverted when it is set to "inverse."

phase shifter on/off
Sets on/off of the phase shifter connected to the direct line. The phase shifter is bypassed with this parameter set to "off," it becomes single stage when the parameters set to 1, and becomes 2-stage with this parameter set to 2.

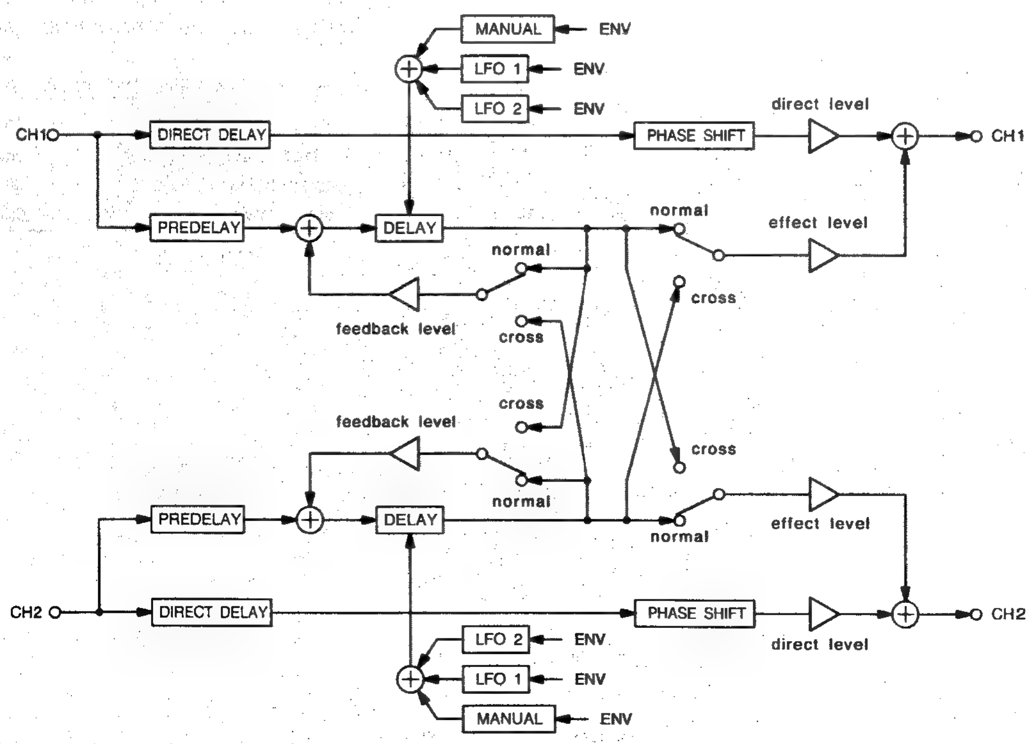
shifting point
Sets the frequency with its phase delayed by 180° (90°). The signal of 12 kHz will be delayed by 180° (90°) when this parameter is set to 0. The frequency the phase of which is delayed by 180° (90°) will become higher as this parameter becomes larger than 0 and will become lower as this parameter becomes smaller than 0.

- The number indicating the phase here is in the case of a 2-stage phase shifter. The figures in parenthesis will be applied in the case of a single-stage phase shifter.

shifting point phase
Sets the phase (±) of the shifting point level.

To be continued ►

Section	Parameter	MIN and MAX
	flanger on/off	on/off
LFO section	LFO 1, 2 frequency (ch1, ch2) LFO 1, 2 freq modulation (ch1, ch2) LFO 1, 2 freq mod phase (ch1, ch2)	0.01 – 40.00Hz 0 – 100.00% normal/inverse
	LFO 1, 2 depth (ch1, ch2) LFO 1, 2 depth modulation (ch1,ch2) LFO 1, 2 depth mod phase (ch1,ch2)	0 – 100.00% 0 – 100.00% normal/inverse
	manual (ch1, ch2) manual modulation (ch1, ch2) manual mod phase (ch1, ch2)	0 – 100.00% 0 – 100.00% normal/inverse
	LFO 1, 2 wave form (ch1, ch2) LFO1, 2 phase (ch1, ch2)	sin/triangle/ special1/special2 0 – 359°
PHASE section	phase shifter on/off (ch1, ch2) shifting point (ch1, ch2) shifting point phase (ch1, ch2)	off/1/2 0 – 99.90% noumal/inverse
DELAY section	direct delay time (ch1, ch2) predelay time (ch1, ch2)	0 – 500.00msec 0 – 500.00msec
	feedback level (ch1, ch2) feedback phase (ch1, ch2) feedback mode (ch1, ch2)	0 – 99.90% normal/inverse normal/cross
LEVEL section	direct level (ch1, ch2) direct phase (ch1, ch2) effect level (ch1, ch2) effect phase (ch1, ch2) effect output mode (ch1, ch2)	0 – 100.00% normal/inverse 0 – 100.00% normal/inverse normal/cross



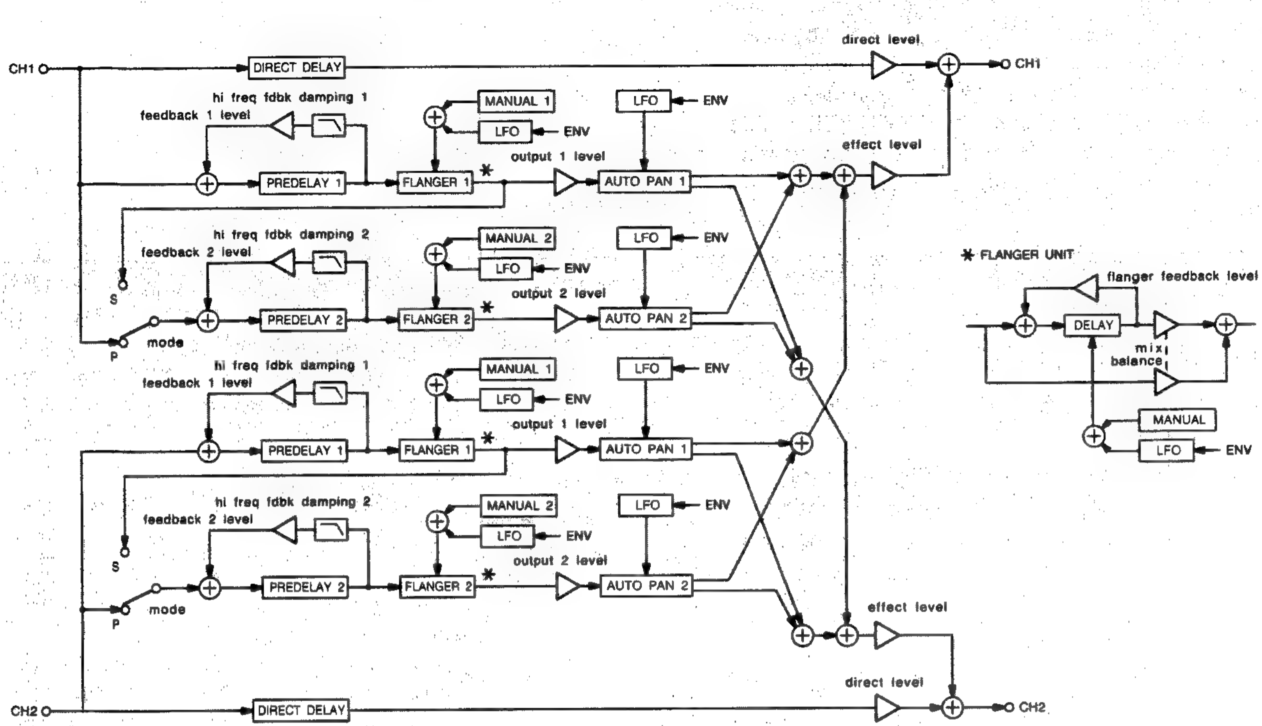
Algorithm 12 Multi Flanger MFL

This is a flanger program provided with 2 flanger units in each channel. Either a parallel or serial connection of 2 flanger units are selectable with "flanger mode."

Parameter description
Many of the parameters in this algorithm are the same as those in Algorithm 11 (stereo flanger). See page 36 for details. Parameters other than these will be described below.
flanger mode
Sets the connection of 2 flanger units to either "parallel" or "serial."
DRY : EFF mix balance
Sets the mixing ratio of effect sound and dry sound in the flanger unit.
autopan frequency
Sets the autopan frequency.
autopan frequency modulation (autopan freq mod)
Sets the effect depth of dynamic "autopan freq mod."
autopan frequency modulation phase (autopan f. mod phase)
Sets the effect of dynamic "autopan f. mod phase." Phase is uninverted when it is set to "normal" and inverted when it is set to "inverse."
autopan depth
Sets the effect depth of autopan. When it is set to 0%, autopan does not operate. The greater this value the greater the width of panning.
autopan phase
Sets the autopan phase angle between two channels.

Section	Parameter	MIN and MAX
	flanger on/off	on/off
	flanger mode	parallel/serial
LFO section	LFO 1, 2 frequency (ch1, ch2)	0.01 – 40.00Hz
	LFO 1, 2 freq modulation (ch1, ch2)	0 – 100.00%
	LFO 1, 2 freq mod phase (ch1, ch2)	normal/inverse
	LFO 1, 2 depth (ch1, ch2)	0 – 100.00%
	LFO 1, 2 depth modulation (ch1,ch2)	0 – 100.00%
	LFO 1, 2 depth mod phase (ch1,ch2)	normal/inverse
	manual 1, 2 (ch1, ch2)	0 – 100.00%
	LFO1, 2 wave form (ch1, ch2)	sin/triangle/ special1/special2
	LFO1, 2 phase (ch1, ch2)	0 – 359°
DELAY section	direct delay time (ch1, ch2)	0 – 200.00msec
	predelay 1, 2 time (ch1, ch2)	0 – 500.00msec
	predelay 1, 2 fdbk level (ch1, ch2)	0 – 99.90%
	predelay 1, 2 fdbk phase (ch1, ch2)	normal/inverse
	hi freq fdbk damping 1, 2 (ch1, ch2)	0.003 – 1.000
FLANGER section	flanger 1, 2 fdbk level (ch1, ch2)	0 – 99.90%
	flanger 1,2 fdbk phase(ch1,ch2)	normal/inverse
	DRY:EFF mix balance 1,2(ch1,ch2)	100:0 – 0:100
	output 1, 2 level (ch1, ch2)	0 – 100.00%
	output 1, 2 phase (ch1, ch2)	normal/inverse
AUTOPAN section	autopan 1, 2 frequency (ch1, ch2)	0.01 – 40.00Hz
	autopan 1, 2 freq mod (ch1, ch2)	0 – 100.00%
	autopan 1, 2 f. mod phase (ch1, ch2)	normal/inverse
	autopan 1, 2 depth (ch1, ch2)	0 – 100.00%
	autopan 1, 2 wave form (ch1, ch2)	sin/triangle/ special1/special2
	autopan 1, 2 phase (ch1, ch2)	0 – 359°
LEVEL section	direct level (ch1, ch2)	0 – 100.00%
	direct phase (ch1, ch2)	normal/inverse
	effect level (ch1, ch2)	0 – 100.00%
	effect phase (ch1, ch2)	normal/inverse

To be continued ►

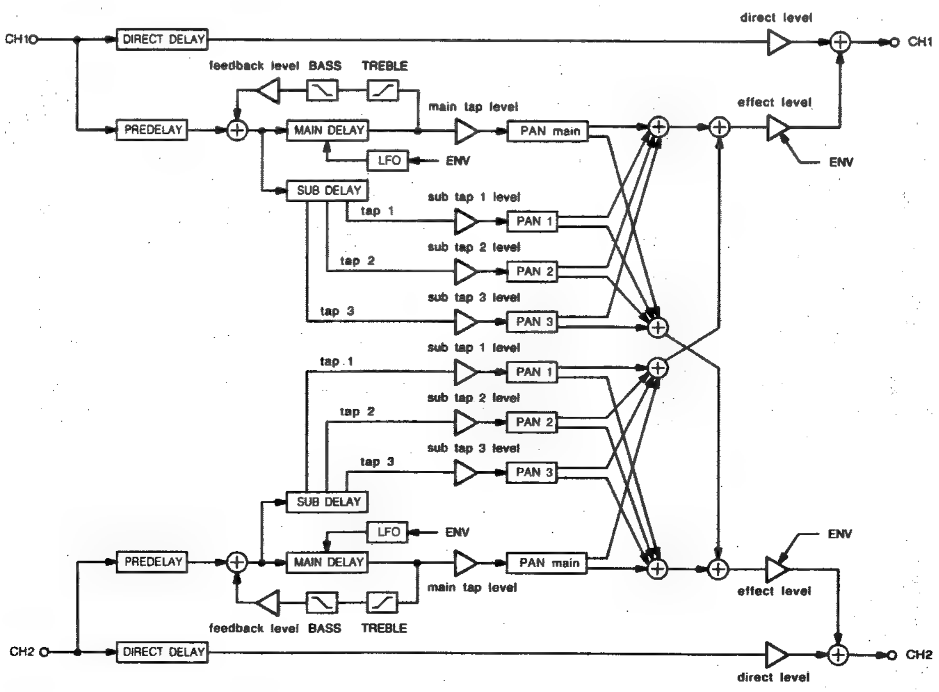


Algorithm 13 Modulation Delay MDL

This algorithm produces a modulation delay effect in which the high frequency range has been accentuated by over-sampling processing (sampling rate: 96 kHz) used in this algorithm. In addition to the main delay line, it has a subdelay line with a 3-tap signal output line which enables the creation of various sounds. Practical sounds can also be produced since the feedback loop has a treble and a bass equalizer.

Section	Parameter	MIN and MAX
	modulation delay on/off	on/off
LFO section	LFO frequency (ch1, ch2)	0.01 – 40.00Hz
	LFO freq modulation (ch1, ch2)	0 – 100.00%
	LFO freq mod phase (ch1, ch2)	normal/inverse
	LFO depth (ch1, ch2)	0 – 100.00%
	LFO depth modulation (ch1, ch2)	0 – 100.00%
	LFO depth mod phase (ch1, ch2)	normal/inverse
	LFO wave form (ch1, ch2)	sin/triangle/ special1/special2
	LFO phase (ch1, ch2)	0 – 359°
DELAY section	direct delay time (ch1, ch2)	0 – 180.00msec
	predelay time (ch1, ch2)	0 – 180.00msec
	main delay time (ch1, ch2)	0 – 500.00msec
	feedback level (ch1, ch2)	0 – 99.90%
	feedback phase (ch1, ch2)	normal/inverse
	main delay tap level (ch1, ch2)	0 – 100.00%
	main delay tap phase (ch1, ch2)	normal/inverse
	main delay tap panpot (ch1, ch2)	0 – 100.00%

Section	Parameter	MIN and MAX
DELAY section	sub delay tap 1 – 3 time (ch1, ch2)	0 – 500.00msec
	sub delay tap 1 – 3 level (ch1, ch2)	0 – 100.00%
	sub delay tap 1 – 3 phase (ch1, ch2)	normal/inverse
	sub delay tap 1 – 3 panpot (ch1,ch2)	0 – 100.00%
EQ section	bass frequency (ch1, ch2)	16Hz – 6.3kHz
	bass level (ch1, ch2)	–12 – +12kB
	treble frequency (ch1, ch2)	400Hz–20.0kHz
	treble level (ch1, ch2)	–12 – +12dB
LEVEL section	direct level (ch1, ch2)	0 – 100.00%
	direct phase (ch1, ch2)	normal/inverse
	effect level (ch1, ch2)	0 – 100.00%
	effect phase (ch1, ch2)	normal/inverse
	effect modulation (ch1, ch2)	0 – 100.00%
	effect mod phase (ch1, ch2)	normal/inverse



To be continued ▶

Algorithm 14 Spiral Modulation SPM

This is an algorithm in which special effects can be created by pitch shift modulation. The effect of raising and lowering the pitch of the input sound can be obtained by "pitch mode" settings. Revolving sound effects can also be created with the panning unit.

Parameter description

pitch mode
Sets the polarity of pitch change. Pitch rising effect with "up" and pitch lowering effects with "down" can be obtained.

pitch LFO frequency
Sets the cycle of pitch change.

pitch LFO frequency modulation (pitch LFO freq mod)
Sets the effect depth of dynamic "pitch LFO freq mod."

pitch LFO frequency modulation phase (pitch LFO f. mod phase)
Sets the effect of dynamic "pitch LFO f. mod phase." Phase is uninverted when it is set to "normal" and inverted when it is set to "inverse."

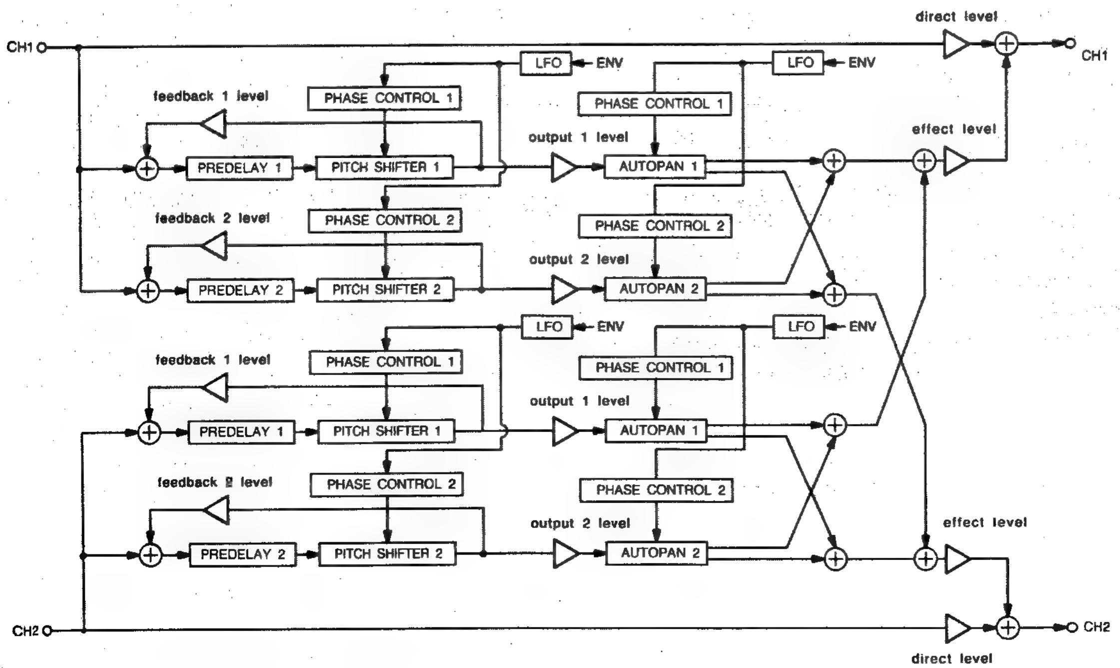
autopan mode
Sets the direction of auto pan movement. Left turn and right turn effects can be obtained with "L turn" and with "R turn."

autopan frequency
Sets the autopan cycle.

autopan frequency modulation (autopan freq mod)
Sets the effect depth of dynamic "autopan freq mod."

autopan frequency modulation phase (autopan freq mod phase)
Sets the effect of dynamic "autopan freq mod phase." Phase is uninverted when it is set to "normal" and inverted when it is set to "inverse."

Section	Parameter	MIN and MAX
	spiral modulation on/off	on/off
PITCH secton	pitch mode (ch1, ch2) pitch LFO frequency (ch1, ch2) pitch LFO freq mod (ch1, ch2) pitch LFO f. mod phase (ch1, ch2) pitch LFO depth (ch1, ch2) pitch LFO phase 1, 2 (ch1, ch2)	up/down 0.01 – 40.00Hz 0 – 100.00% normal/inverse 0 – 100.00% 0 – 359°
DELAY section	predelay1, 2 time (ch1, ch2)	0 – 300.00msec
	feedback 1, 2 time (ch1, ch2) feedback 1, 2 phase (ch1, ch2) output 1, 2 level (ch1, ch2) output 1, 2 phase (ch1, ch2)	0 – 99.90% normal/inverse 0 – 100.00% normal/inverse
AUTOPAN section	autopan mode (ch1, ch2) autopan frequency (ch1, ch2) autopan freq mod (ch1, ch2) autopan freq mod phase(ch1,ch2) autopan depth (ch1, ch2) autopan phase 1, 2 (ch1, ch2)	L turn/R turn 0.01 – 40.00Hz 0 – 100.00% normal/inverse 0 – 100.00% 0 – 359°
LEVEL section	direct level (ch1, ch2) direct phase (ch1, ch2) effect level (ch1, ch2) effect phase (ch1, ch2)	0 – 100.00% normal/inverse 0 – 100.00% normal/inverse



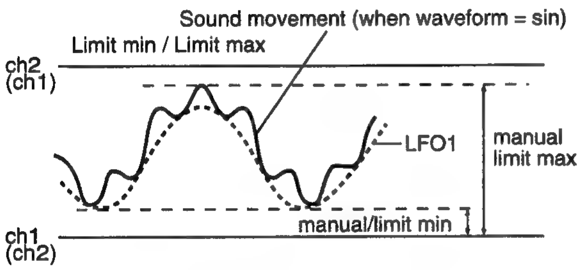
Algorithm 15 Stereo Panner SPA

This is an algorithm in which three types of panning effects can be obtained by "mode" settings. Auto panning multiplexed by two LFOs will be carried out in "LFO" mode, auto panning by output waveform from the envelope block will be carried out in "ENV" (envelope) mode, and panning by manual setting only will be performed in "off" mode.

Parameter description

manual limit min/max

Sets the movement width of the panning. Panning by manual modulation operates within the range between 100% width of "manual limit min" and that of "manual limit max". Moreover, the position of "manual limit min" becomes 0% level set in the envelope block and that of "manual limit max" becomes +100% level set in the envelope block, when "ENV" (envelope) is selected in the "mode" parameter.



LFO1 : LFO2 mix balance

Sets the mixing ratio of LFO1 and LFO2. LFO1 will have the maximum amplitude at 100 : 0 and LFO2 will have the maximum amplitude at 0 : 100.

LFO start point/LFO step

This is a parameter that sets the initial status of the sound locating. The sound moves one LFO step each time a trigger is input. The available range of "LFO step" is from 1° to 360° (the sound returns to the initial position).

trigger select

Selects the trigger to start the LFO. The volume level of input signal to each channel is the trigger for "signal" and "ch1 (available for "trigger 1, 2 select ch2" only)" and MIDI note on information is the trigger for "MIDI." If "key" is selected, the "key trigger" of the system block will be the trigger. If "off" is selected, sound continues to move.

trigger threshold

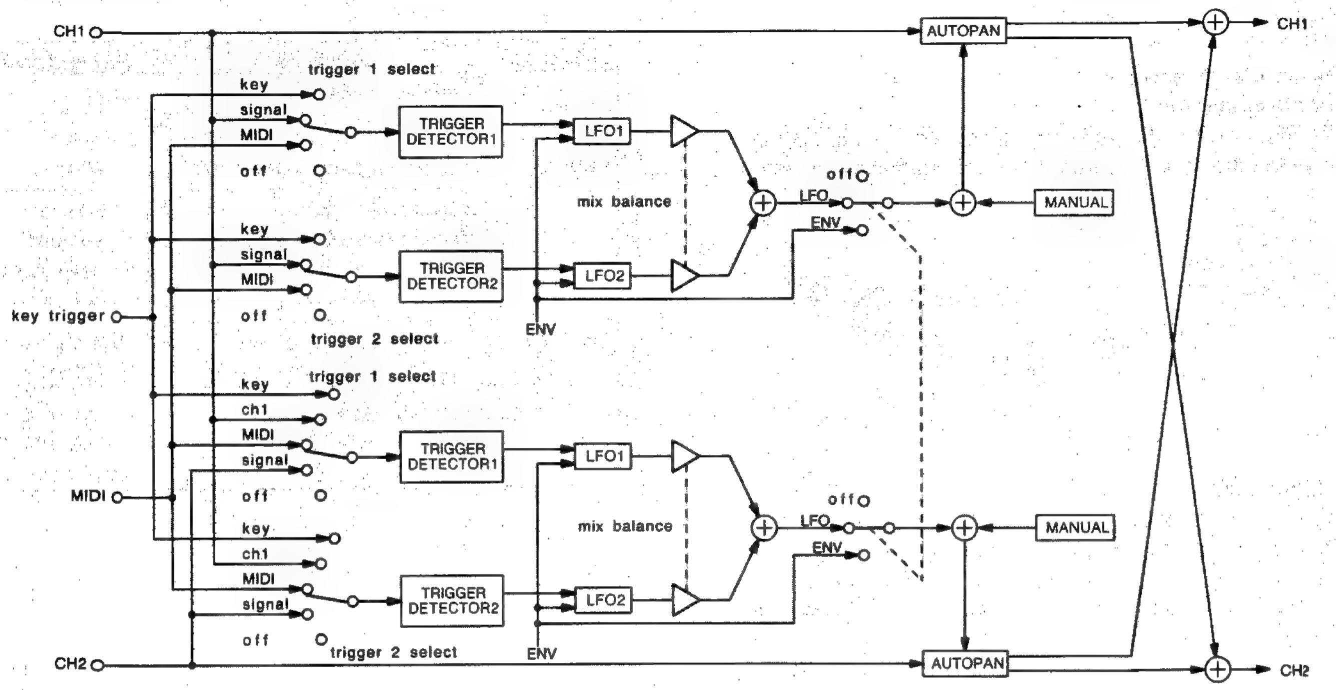
Auto panning starts if a signal greater than the value set by this parameter is input when "trigger select" is set to "signal" or "ch1."

Note

Trigger input is not possible while sound is moving with a trigger.

Section	Parameter	MIN and MAX
	stereo panner on/off	on/off
	mode	off/LFO/ENV
	manual limit min (ch1,ch2)	0 – 100.00%
	manual limit max (ch1,ch2)	0 – 100.00%
LFO section	LFO 1, 2 frequency (ch1,ch2)	0.01 – 40.00Hz
	LFO 1, 2 freq modulation (ch1,ch2)	0 – 100.00%
	LFO 1, 2 freq mod phase (ch1,ch2)	normal/inverse
	LFO 1, 2 wave form (ch1,ch2)	sin/triangle/special1/special2
	LFO 1: LFO2 mix balance (ch1,ch2)	100:0 – 0:100
	LFO 1, 2 start point (ch1,ch2)	0 – 359°
	LFO 1,2 step (ch1,ch2)	1 – 360°
TRIG.section	trigger 1, 2 select (ch1,ch2)	off/MIDI/signal/key/ch1
	trigger 1, 2 threshold (ch1,ch2)	0 – 100.00%

To be continued ▶



Algorithm 16 Haas Panner HPA

This is an auto panning algorithm using the Haas effect. Natural panning with no uncomfortable feeling when the sound image moves from one channel to another even while listening with headphones.

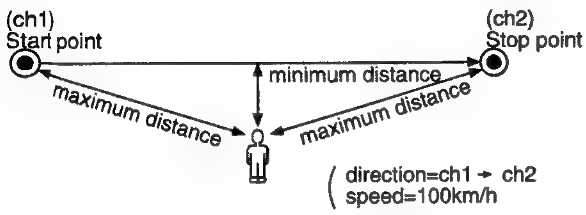
Parameter description

The parameters and block diagrams are the same as those in Algorithm 15 (stereo panner). See page 42 for details.

Algorithm 17 Doppler DOP

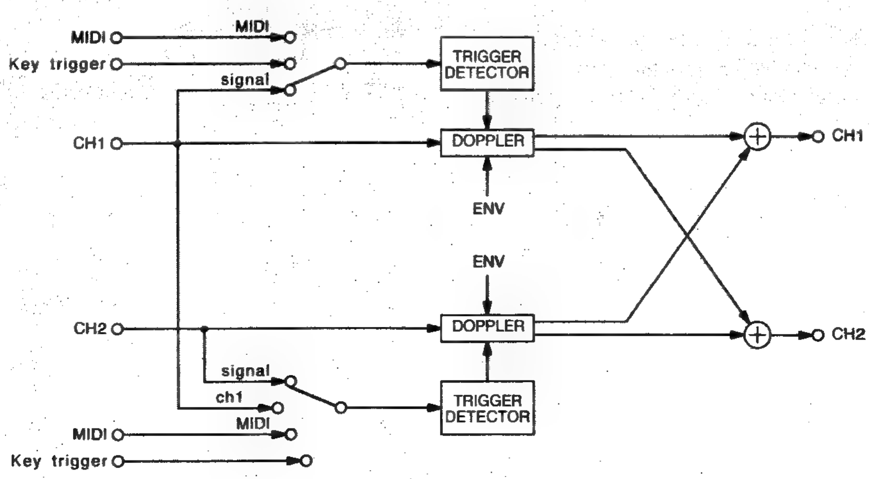
This algorithm provides a doppler effect.

Parameter description
locating section
As shown in the diagram, this parameter sets the condition in which the sound source moves in relation to the listener.



trigger select
Selects the trigger to start the LFO. The volume level of input signal to each channel is the trigger for "signal" and "ch1 (available for "trigger select ch2" only)" and MIDI note on information is the trigger for "MIDI." If "key" is selected, the setting of "key trigger" of the system block will be applied as the trigger. If "off" is selected, sound continues to move.
trigger threshold
Doppler effect starts if a signal greater than the value set by this parameter is input when "trigger select" is set to "signal" or "ch1."
ambience
Sets the sound field impression.
If this value is increased, the sound field impression will expand.

Section	Parameter	MIN and MAX
	doppler on/off	on/off
LOCATING section	maximum distance (ch1, ch2) minimum distance (ch1, ch2)	0 – 300m 0 – 300m
	speed (ch1, ch2) speed modulation (ch1, ch2) speed mod phase (ch1, ch2) direction (ch1, ch2)	0 – 500km/h 0 – 100.00% normal/inverse ch1→ 2/ch2→ 1
TRIG. section	trigger select (ch1, ch2) trigger threshold (ch1, ch2)	MIDI/signal/key/ch1 0 – 100.00%
LEVEL section	ambience (ch1, ch2) output level (ch1, ch2) output phase (ch1, ch2)	0 – 100.00% 0 – 100.00% normal/inverse



To be continued ►

Algorithm 18 Vibrato + Tremolo VIB

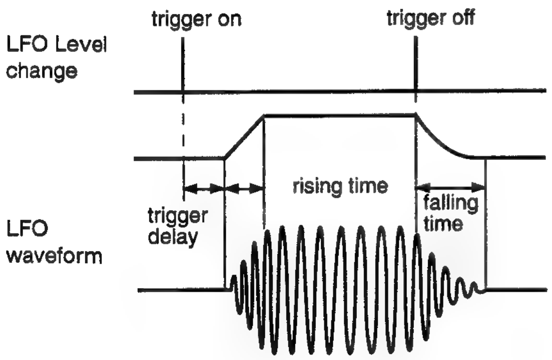
This algorithm produces a vibrato and tremolo effect in which the high frequency range has been accentuated by oversampling processing (sampling rate: 96 kHz) used in this algorithm.

Parameter description

trigger select
Selects the trigger to start the LFO. The volume level of input signal to each channel is the trigger for "signal" and "ch1 (available for "trigger select ch2" only)" and MIDI note on information is the trigger for "MIDI." If "key" is selected, the "key trigger" of the system block will be the trigger. If "off" is selected, sound continues to move.

trigger threshold
Vibrato and tremolo start if a signal greater than the value set by this parameter is input when trigger select is set to "signal" or "ch1."

trigger delay/rising time/falling time
Sets the time which the trigger actually takes to become effective after the input signal exceeding the "trigger threshold" level is triggered on, when "trigger select " is set to "signal" or "ch1."



The LFO output level is controlled in the TRIG.(trigger) section. As shown in the figure above, the LFO output level changes in accordance with the values of "trigger delay" and "rising time" after trigger-on. The LFO output level attenuates in accordance with the value of "falling time" after trigger-off. "trigger delay" is effective only during trigger-on.

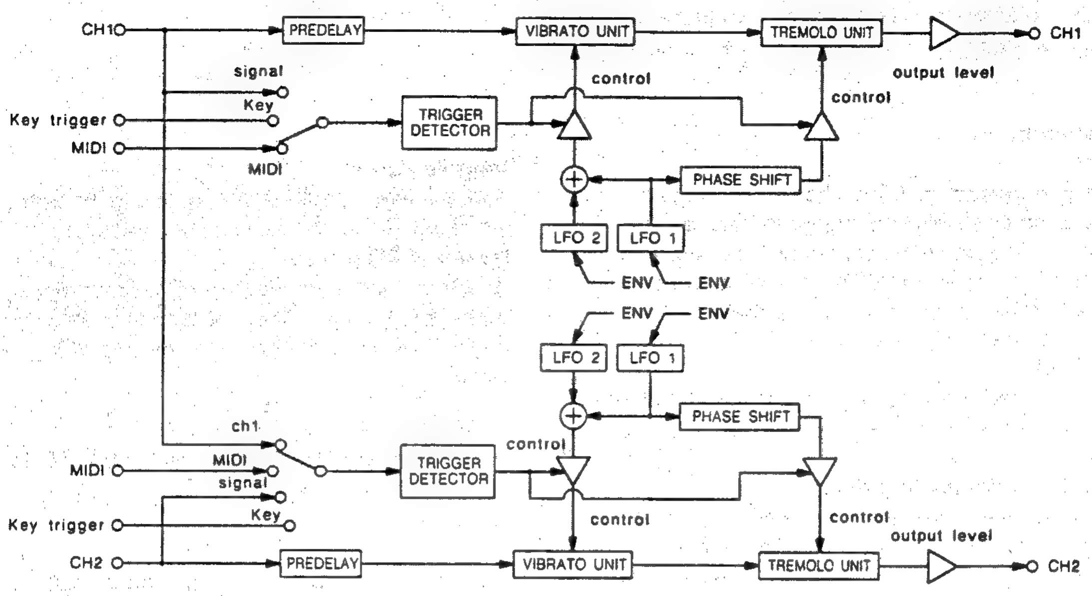
tremolo depth
Sets the effect depth of the tremolo in the tremolo unit. The effect increases as this value becomes larger.

tremolo LFO phase
Sets the phase angle of the LFO of the tremolo unit. A panning-like effect can be obtained by shifting the phase angle between channels even when inputting monaural signals.

Note
The same value as in "LFO1 frequency" of vibrato will be set for "LFO frequency" of tremolo.

Section	Parameter	MIN and MAX
	vibrato + tremolo on/off	on/off
LFO section	LFO 1, 2 frequency (ch1, ch2) LFO 1, 2 freq modulation (ch1, ch2) LFO 1, 2 freq mod phase (ch1, ch2)	0.01 – 40.00Hz 0 – 100.00% normal/inverse
	LFO 1, 2 depth (ch1, ch2) LFO 1, 2 depth modulation (ch1,ch2) LFO 1, 2 depth mod phase(ch1,ch2)	0 – 100.00% 0 – 100.00% normal/inverse
	LFO 1, 2 wave form (ch1, ch2) LFO 1, 2 phase (ch1, ch2)	sin/triangle 0 – 359°
TRIG. section	trigger select (ch1, ch2) trigger threshold (ch1, ch2) trigger delay (ch1, ch2) rising time (ch1, ch2) falling time (ch1, ch2)	MIDI/signal/key/ch1 0 – 100.00% 0 – 10.00sec 0 – 10.00sec 0 – 10.00sec
TREMOLO section	tremolo depth (ch1, ch2) tremolo LFO phase (ch1, ch2)	0 – 100.00% 0 – 359°
DELAY section	predelay time (ch1, ch2)	0 – 500.00msec
LEVEL section	output level (ch1, ch2) output phase (ch1, ch2)	0 – 100.00% normal/inverse

Modulation Block



To be continued ▶

Algorithm 19 Ring Modulator RNG

This is an algorithm in which feedback delay is connected to the ring modulator. Sound effects such as bells, chimes, explosions and gunfire can be produced by combining two signals.

Parameter description

OSC frequency
Sets the oscillating frequency of a sine wave oscillator.

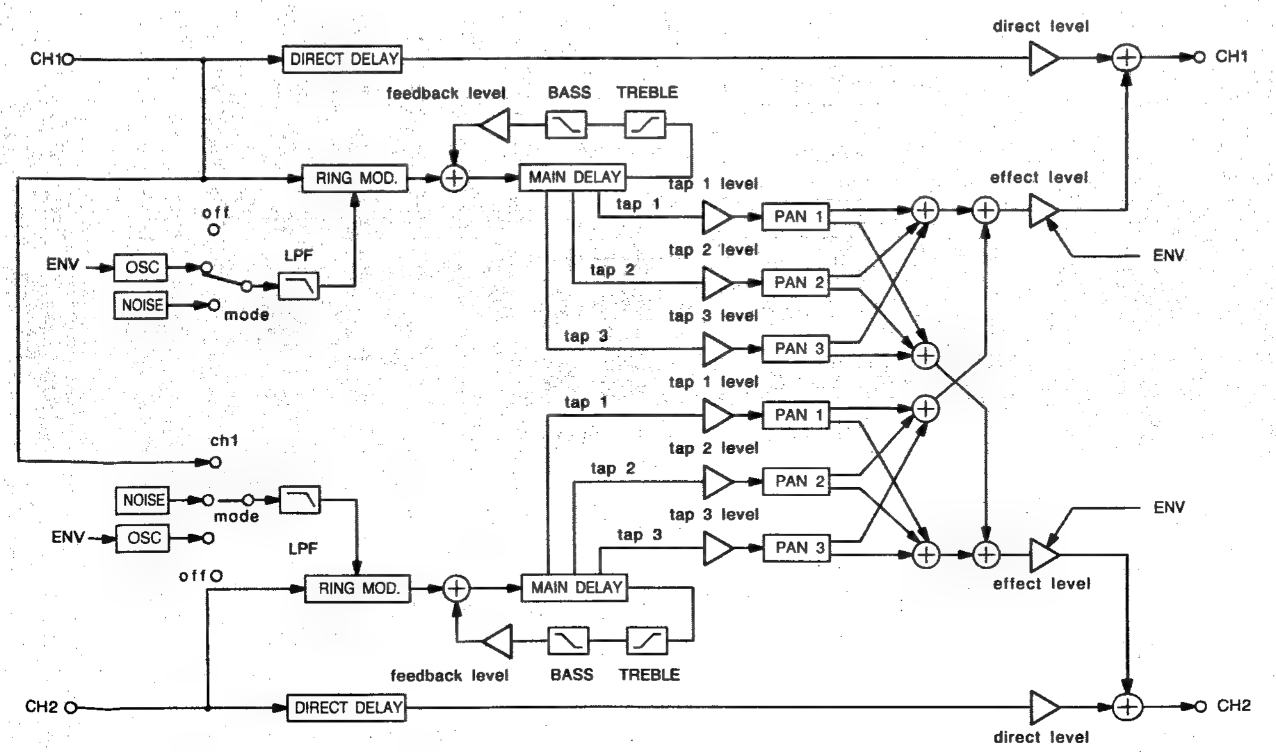
OSC frequency modulation (OSC freq modulation)
Sets the effect depth of dynamic "OSC freq modulation."

OSC frequency modulation phase (OSC freq mod phase)
Sets the effect of dynamic "OSC freq mod phase." Phase is uninverted when it is set to "normal" and inverted when it is set to "inverse."

ring modulator mode
Selects the carrier used in the ring modulator. Sine waves can be selected with "osc" and white noise with "noise." When it is set to "off," the ring modulator unit will be passed through and only the delay unit will be used. Furthermore, "ch1" can be selected as the carrier for channel 2.

LPF cutoff frequency
Sets the cutoff frequency when applying an LPF (low pass filter) to the carrier signal. When it is set to "thru," the LPF will be passed through.

Section	Parameter	MIN and MAX
	ring modulator on/off	on/off
OSC. section	OSC frequency (ch1, ch2) OSC freq modulation (ch1, ch2) OSC freq mod phase (ch1, ch2) OSC phase (ch1, ch2) ring modulator mode (ch1, ch2)	0.05 – 3000.00Hz 0 – 100.00% normal/inverse 0 – 359° off/osc/noise/ch1
DELAY section	direct delay time (ch1, ch2) main delay time (ch1, ch2) feedback level (ch1, ch2) feedback phase (ch1, ch2) main delay tap 1 – 3 time (ch1, ch2) main delay tap 1 – 3 level (ch1, ch2) main delay tap 1 – 3 phase(ch1,ch2) main delay tap 1–3 pan (ch1,ch2)	0 – 350.00msec 0 – 1000.00msec 0 – 99.90% normal/inverse 0 – 1000.00msec 0 – 100.00% normal/inverse 0 – 100.00%
EQ section	bass frequency (ch1, ch2) bass level (ch1, ch2) treble frequency (ch1, ch2) treble level (ch1, ch2)	16Hz – 6.3kHz –12 – +12dB 400Hz – 20.0kHz –12 – +12dB
	LPF cutoff frequency (ch1, ch2)	100Hz – 6.3kHz/thru
LEVEL section	direct level (ch1, ch2) direct phase (ch1, ch2) effect level (ch1, ch2) effect phase (ch1, ch2) effect modulation (ch1, ch2) effect mod phase (ch1, ch2)	0 – 100.00% normal/inverse 0 – 100.00% normal/inverse 0 – 100.00% normal/inverse

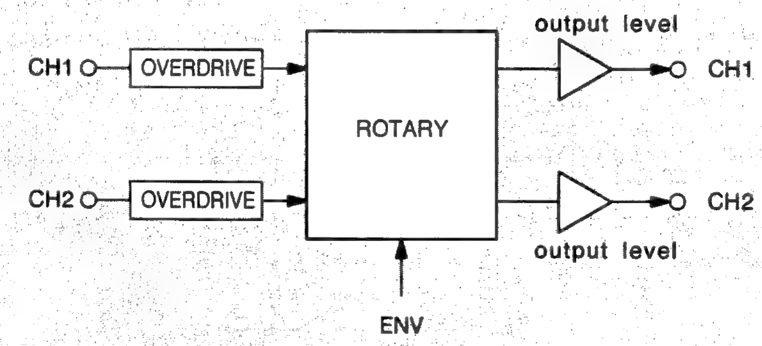


Algorithm 20 Rotary Speaker RTY

This is an algorithm for simulating the sound of rotary speakers.

Parameter description
horn speed (fast/slow)
Sets the horn rotation speed when set to "fast" or "slow."
rotor speed (fast/slow)
Sets the rotor rotation speed when set to "fast" or "slow."
rising time (horn/rotor)
Sets the time required for changing the rotation speed from "slow" to "fast."
falling time (horn/rotor)
Sets the time required for changing the rotation speed from "fast" to "slow."
drive
Adjusts the overdrive level. As the value for "drive" increases, distortion will increase.
level
Adjusts the output level of the DRIVE section.

Section	Parameter	MIN and MAX
	rotary speaker on/off	on/off
SPEED section	speed select	fast/slow
	horn speed (fast)	1.00 – 20.00Hz
	horn speed (slow)	0.05 – 5.00Hz
	horn speed modulation	0 – 100.00%
	horn speed mod phase	normal/inverse
	rotor speed (fast)	1.00 – 20.00Hz
	rotor speed (slow)	0.05 – 5.00Hz
	rotor speed modulation	0 – 100.00%
	rotor speed mod phase	normal/inverse
	rising time (horn)	0.01 – 10.00sec
	falling time (horn)	0.01 – 10.00sec
	rising time (rotor)	0.01 – 10.00sec
	falling time (rotor)	0.01 – 10.00sec
	horn depth	0 – 100.00%
	rotor depth	0 – 100.00%
DRIVE section	drive	0 – 100.00%
	level	0 – 100.00%
LEVEL section	ambience	0 – 100.00%
	horn : rotor mix balance	100:0 – 0:100
	output level	0 – 100.00%



Post-effect Block

This block processes signals from the modulation block and outputs to the output block. Signal processing in this block uses 4 different algorithms (excluding Algorithm 0) according to the preset memory

When editing a preset memory, first confirm the type of algorithm used in the preset memory. Parameters also differ according to the algorithm.

Algorithm 0 Effect Off OFF

Post-effect block will be passed through so that no effect can be obtained in this block.

Algorithm 1 Stereo Equalizer SEQ

See page 13 for parameters and block diagram.

Algorithm 2 Stereo Exciter + Stereo EQ SXE

See page 14 for parameters and block diagram.

Algorithm 3 Dynamic Exciter DEX

See page 15 for parameters and block diagram.

Algorithm 4 Gate GTE

See page 16 for parameters and block diagram.

Envelope Block

This block is composed of the envelope follower and envelope generator and its output is used for real time control of the modulation block parameters. Signal processing in this block uses 3 different algorithms (excuding Algorithm 0) according to the preset memory. Basic waveforms are formed in this block for overall level and phase adjustments in the modulation block.

When editing a preset memory, first confirm the type of algorithm used in the preset memory. Parameters also differ according to the algorithm.

Algorithm 0 — Effect Off — OFF

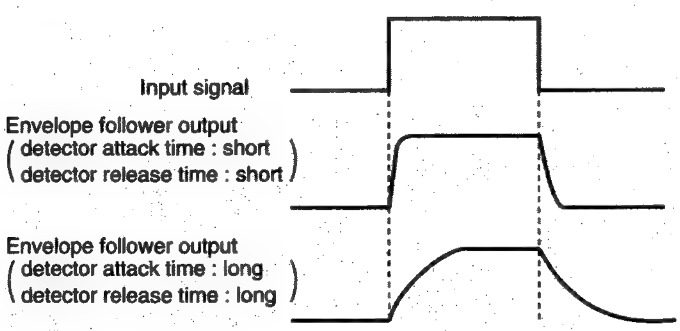
Envelope block is not used. The output of the envelope block will be 0.

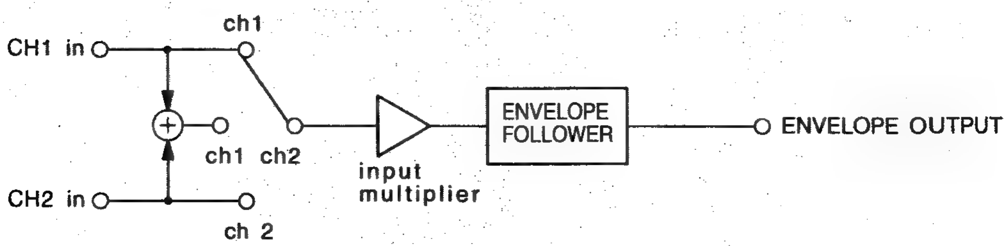
Algorithm 1 — Envelope Follower — EVF

This algorithm outputs a waveform following the input signal and is used to change the modulation block parameters by changing the volume.

Parameter description
detector attack time, detector release time
The shorter the "detector attack time" is, the quicker the envelope follower output is, following the rising of the input signal. On the other hand, the shorter the "detector release time" is, the quicker the envelope follower output is, following the release of the input signal.
input multiplier
Amplifies the input signal. It is selectable from "x1," "x2" and "x4."

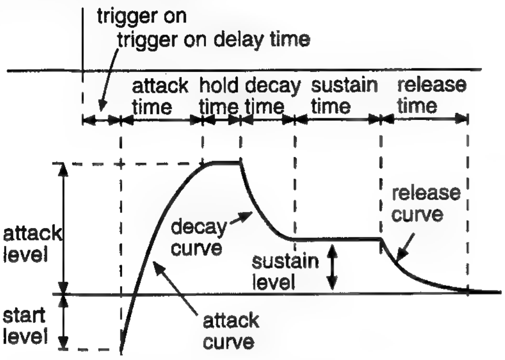
Parameter	MIN and MAX
envelope follower on/off	on/off
signal select	ch1/ch2/ch1+ch2
input multiplier	x1/x2/x4
detector attack time detector release time	0 – 500msec 1 – 5000msec





Algorithm 2 Envelope Generator 1 EG1

This algorithm outputs an envelope waveform with the input level or MIDI note on information as the trigger. In this algorithm, "sustain time" can be adjusted.

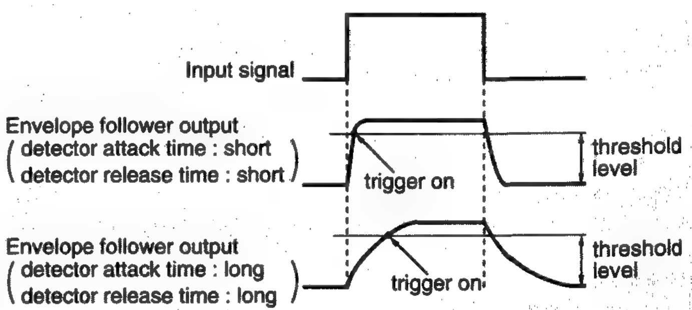


Parameter	MIN and MAX
envelope generator 1 on/off	on/off
trigger select	ch1/ch2/ch1+ch2/MIDI/key
threshold level	0 – 100%
trigger mask time trigger on delay time attack time hold time decay time sustain time release time	0 – 60.00sec, auto 0 – 10.00sec 0 – 10.00sec 0 – 10.00sec 0 – 10.00sec 0 – 10.00sec 0 – 10.00sec
start level attack level sustain level	–100 – +100% –100 – +100% –100 – +100%
attack curve decay curve release curve	linear/curve 1/curve 2 linear/curve 1/curve 2 linear/curve 1/curve 2
detector attack time detector release time	0 – 500msec 1 – 5000msec

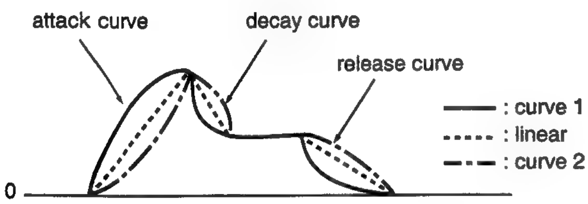
Parameter description

detector attack time, detector release time and threshold level

When triggering with volume level in EG1 (when "trigger select" = "ch1/ch2/ch1+ch2"), envelope follower processing is performed on the input signal and the results are compared with the "threshold level." If the envelope follower output is above the "threshold level" setting, the trigger will go on. Similar to Algorithm 1, "detector attack time" and "detector release time" are time constants relative to changing input signals.



attack curve, decay curve, release curve

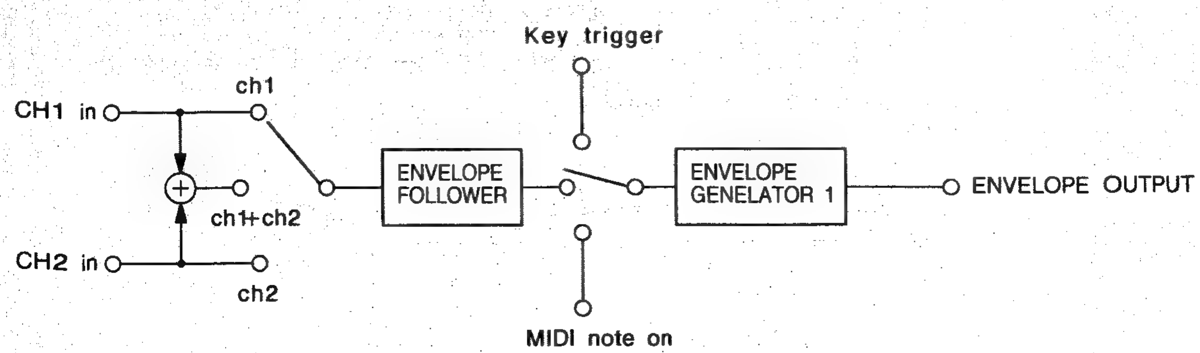


trigger masking time

Sets the time from a trigger acceptance until the next trigger acceptance. If it is set to "auto," triggers will not be accepted until release ends.

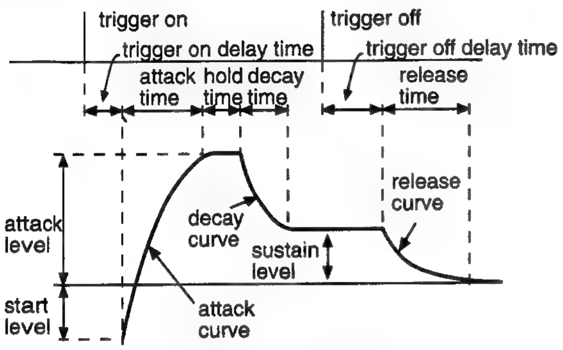
trigger select

If "key" is selected, the setting of "key trigger" of the system block will be applied as the trigger.



Algorithm 3 Envelope Generator 2 EG2

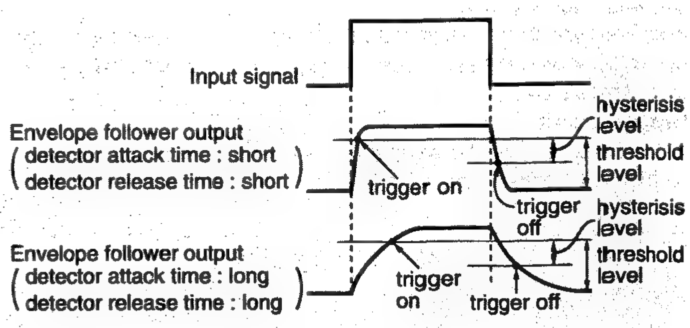
This algorithm outputs an envelope waveform with the input level or MIDI note on information as the trigger. This is a type in which release is started when the input level drops below the level specified by "hysteresis level" or when MIDI note off (or note on with the velocity set to 0) is used as the trigger-off signal.

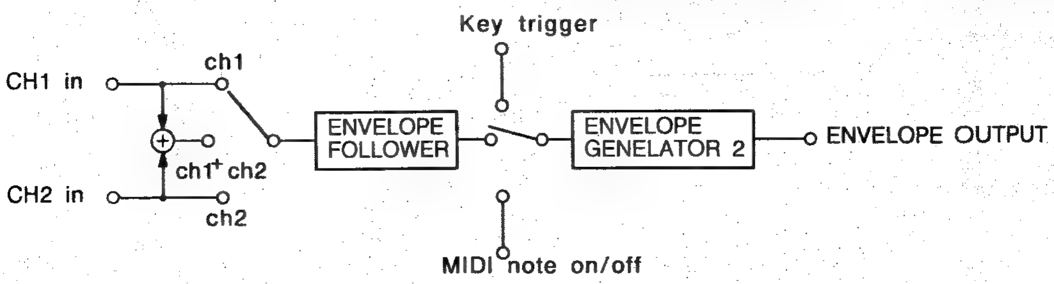


detector attack time, detector release time, threshold level
When applying the trigger in EG2 (when "trigger select "= "ch1/ch2/ch1+ch2") with the volume level, envelope follower processing is executed on the input signal. When the volume level is above the "threshold level" setting, the trigger goes on and is goes off when the volume level is below the level specified by the "hysteresis level." Similar to Algorithm 1, "detector attack time" and "detector release time" are time constants relative to the change of the input signal.
attack curve, decay curve and release curve
See page 51.
trigger masking time
Sets the time from a trigger acceptance until the next trigger acceptance.
release time
If a trigger-off signal is received, the signal is delayed functioning for the time set with "trigger off delay time," and then decreases down to 0 level for the time set with "release time."

trigger select
If "key" is selected, the setting of "key trigger" of the system block will be applied as the trigger.
hysteresis level
This is indicated in proportion to "threshold level." For example, if the "hysteresis level" is set to 0%, the trigger-on and trigger-off levels will be the same (that is, "threshold level"). If the "hysteresis level" is set to 50%, the trigger-off level will become 50% of the trigger-on level.

Parameter	MIN and MAX
envelope generator 2 on/off	on/off
trigger select	ch1/ch2/ch1+ch2/MIDI/key
threshold level hysterisis level	0 – 100% 0 – 100%
trigger mask time trigger on delay time trigger off delay time attack time hold time decay time release time	0 – 60.00sec 0 – 10.00sec 0 – 10.00sec 0 – 10.00sec 0 – 10.00sec 0 – 10.00sec 0 – 10.00sec
start level attack level sustain level	–100 – +100% –100 – +100% –100 – +100%
attack curve decay curve release curve	linear/curve 1/curve 2 linear/curve 1/curve 2 inear/curve 1/curve 2
detector attack time detector release time	0 – 500msec 1 – 5000msec





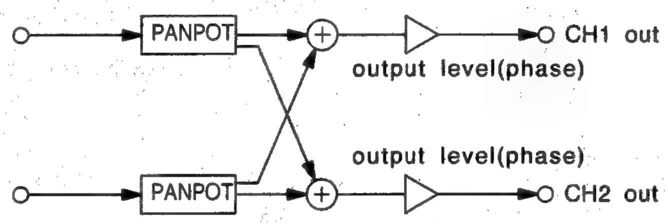
Output Block

To be continued ►

This block receives signals from the post-effect block and gives them panpot, level and phase processing. The control here is useful in adjusting the level difference in each memory.

Parameters	MIN and MAX
output level (ch1, ch2)	0 – 100%
output phase (ch1, ch2)	normal/inverse
output panpot (ch1, ch2)	0 – 100%

Note
If "output panpot" is set to 0 %, signals input for each channel are output to the same channel.
If it is set to 100 %, input signals for ch1 are output to ch2 and vice versa.



Other Blocks

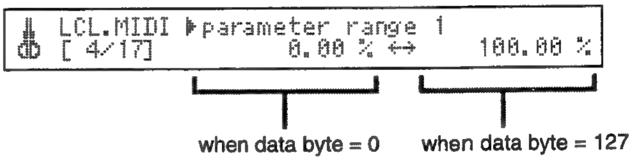
LCL. MIDI (Local MIDI) Block

This is a MIDI parameter setting block that can be set in each memory.

Parameter	Meaning
control no.1 - 4	Sets MIDI control change number. Selects from off, 0 - 120 or from key velocity, channel key pressure and note number.
parameter block 1 - 4	Selects signal processing block for control with numbers set in control no. 1 - 4.
parameter name 1 - 4	Selects parameter for control with numbers set in control no. 1 - 4.
parameter range 1 - 4	Specifies control range of parameters set in parameter name 1 - 4.

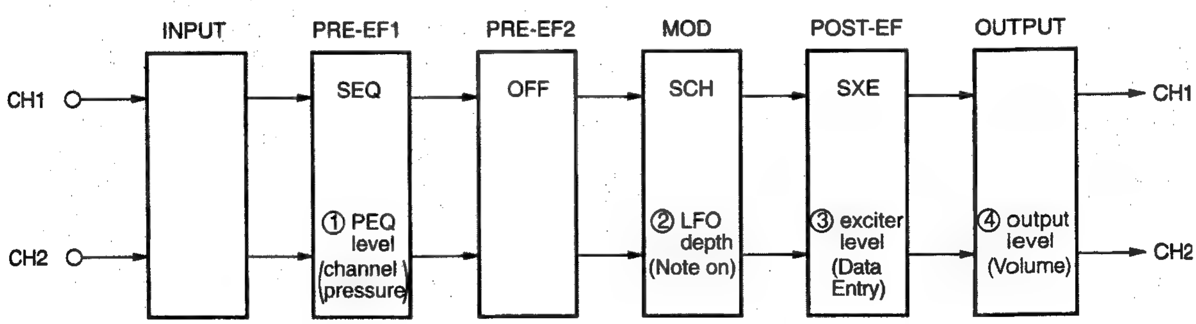
parameter range

[Example of selecting the parameter range]



To enter the parameter range selecting mode
Press the ENTER button when the parameter selecting screen is displayed and then set the parameter value with MIDI control change and note on data bytes setting to 0.
Press the ENTER button once again and set the parameter value with the data byte setting to 127. Then again press the ENTER button, the display will return to the parameter selecting screen.

Applications for LCL. MIDI



1. Channel Pressure (After touch) controls the PEQ level of Stereo EQ.
2. MIDI note no. data controls the LFO depth. Playing on a higher key possibly enhances the depth.
3. Data Entry (controller no. 6) controls the exciter level. The data entry volume on the synthesizer performs fine adjustment of the exciter level.
4. Volume (controller no. 7) controls the output level. A foot volume connected to the synthesizer controls the effect volume.

To be continued ▶

System Block

This block specifies the operating environment of the DPS-M7.

Parameter	Meaning
input mode	Selects the input mode (stereo/mono). In monaural mode, only the INPUT CH1 terminals are available.
auto help	Selects whether HELP message is to be displayed automatically or not.
load form	Selects auto load or enter load. auto load — a memory is automatically called up when you dial the memory number in load mode. enter load — a memory is not called up until you dial the memory number in load mode and then press the ENTER button.
load time	Sets the memory access time after the number is changed in auto load mode. Available range is from 200 msec to 1000 msec.
dial sensitivity	Adjusts the sensitivity of the dial within the range of 1 to 12. The dial sensitivity increases as the numbers become smaller.
unit (time)	Sets the unit for time information such as predelay time. Select either "word" or "msec." • "word" indicates the number of samples.
unit (level)	Sets the units for level information. Select either "%" or "dB."
unit (q)	Sets the unit for q of EQ. Select either "q" or "oct".
remote ch	Sets the remote channel. Select from 1 to 15 channels.
remote baud rate	Selects the remote baud rate. Select from 9600 to 31250 bps.
clock set	Sets the calendar and time. Cursor moves at a press of the EDIT button and clock can be confirmed when the parameter menu is displayed.
user's name	Enter your name. Cursor moves at a press of the EDIT button.
date of birth	Enter your birth date. Cursor moves at a press of the EDIT button.
key protect	This function will no longer accept key operations. This function is to prevent misoperation by someone else. To release "key protect," press both the EDIT and ENTER buttons at the same time and turn the dial counterclockwise.
key trigger	If "key" is selected in "trigger select" parameters of the algorithms for the modulation block and envelope block, trigger-on will occur if the ENTER button is pressed and trigger-off (only in algorithms where trigger-off is effective) will occur when the button is released.
battery check	Checks the battery for maintaining the user memory.
version check	Software version can be verified.

Memory Block

This is the block for editing the user memory.

Parameter	Meaning
memory compare	For comparative listening with original memory. The following selections are available. •"edit/memory" •"edit/parameter" •"edit/parameter/memory" •"edit/parameter/block/memory" "edit" : Normal editing mode "parameter" : Only the currently displayed parameter has a value before changing. "block" : Only the currently displayed block is original "memory" : Original data
move	Moves a specified user memory to another number.
copy	Copies a specified preset memory and/or user memory to another number.
delete	Deletes a specified user memory.
exchange	Exchanges specified user memories
remaining area	Indicates the remaining area of the user memory.
protect U1 - U256	Turns on/off the memory protection for a specified user memory.

SYS. MIDI (System MIDI) Block

This block specifies MIDI operating environment of the DPS-M7.

Parameter	Meaning
MIDI on/off	Turn on/off receiving of MIDI data (excluding system exclusive message). ("on" when the power is on)
MIDI omni	Sets MIDI OMNI on/off. MIDI data is received regardless of MIDI channels when OMNI is "on."
MIDI ch	Sets the MIDI channel. Select from 1 to 16 ch.
bulk dump transfer	Transfers memory data system information through MIDI. The following information can be transformed. •"all" (all user memories, system information, MIDI information) •"all user's memory" (all user's memories) •"system" (all system information) •"all MIDI" (all MIDI information) •"user's memory" (one specified user memory)
bulk dump receive on/off	Turn on/off receiving of the bulk dump data. ("off" when the power is on)
program change no. 1-128	Sets the memory numbers corresponding to MIDI program change numbers 1 to 128. Select from P1 to P100, U1 to U256, BYPASS.

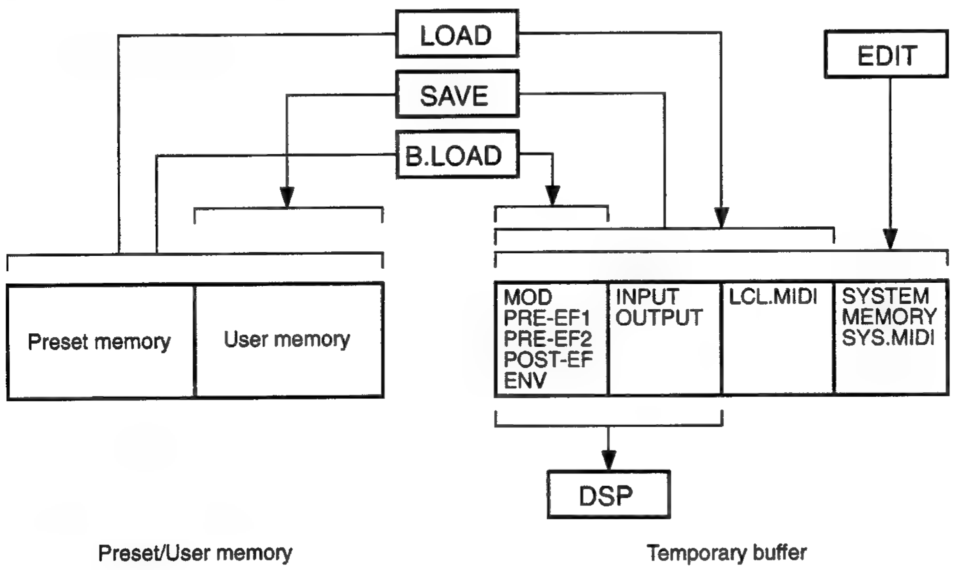
To be continued ▶

B. LOAD (Block Load) Block

This block allows you to load signal processing blocks partially from another memory (preset or user) in the temporary buffer being edited. This function corresponds to the pre-effect 1, pre-effect 2, modulation, post-effect and envelope blocks.

Parameters	**Meanings**
MOD block load	Loads the modulation block of another preset or user memory in the modulation area of the temporary buffer.
PRE-EF 1 block load	Loads one block from the pre-effect 1, pre-effect 2, and post-effect blocks of another preset or user memory in the pre-effect 1 area of the temporary buffer.
PRE-EF2 block load	Loads one block from the pre-effect 1, pre-effect 2, and post-effect blocks of another preset or user memory in the pre-effect 2 area of the temporary buffer.
POST-EF block load	Loads one block from the pre-effect 1, pre-effect 2, and post-effect blocks of another preset or user memory in the post-effect area of the temporary buffer.
ENV block load	Loads the envelope block of another preset or user memory in the envelope area of the temporary buffer.

Relation between memory and temporary buffer



Temporary buffer

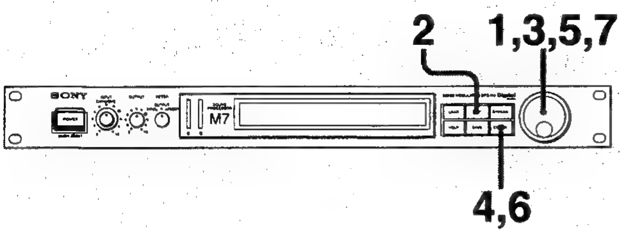
To copy a preset memory and a user memory data in the temporary buffer is called to "load". To Edit the data in the temporary buffer and to write the results in the user memory is called to "save". Data in the temporary buffer is transferred to the DSP (Digital Signal Processor) and is reflected in the sound.

Notes

- When operating the block load function, the LCL. MIDI parameters are initialized as follows.
  control no. : off
  parameter block : INPUT
  parameter name: input level sync
  parameter range: 0 – 100.00%

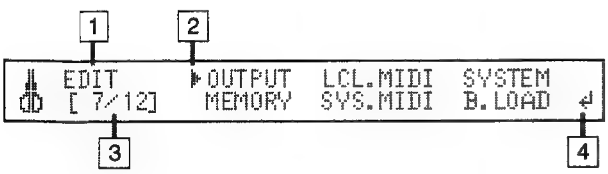
- Block loading is used, like other parameters, to change the temporary buffer being currently edited. In other words, it is not to directly overwrite the contents of the preset memory or the user memory saved. It is necessary to save when desiring to preserve the edited effect.

B. LOAD Operation



1. Call the memory number to be changed.

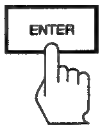
2. Press the EDIT button so that the block selecting screen will be displayed.

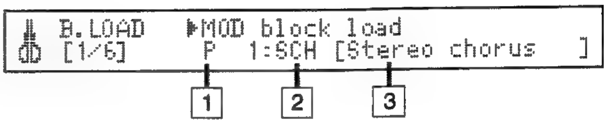


1. EDIT mode indication
2. Displayed on the left side of the currently selected item.
3. This shows there are 12 selections and the seventh of these is selected.
4. This means no more items following. If there are more items following , "→" appears.

3. Turn the dial and select B. LOAD.

4. Press the ENTER button.

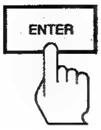


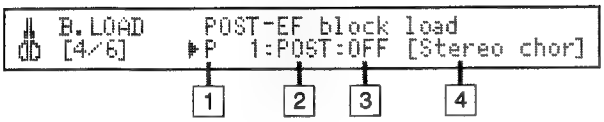


1. Preset memory number or user memory number
2. Algorithm name
3. Preset memory or user memory name (first 16 characters)

5. Turn the dial and select the block to be copied.

6. Press the ENTER button.





1 Preset memory number or user memory number
2 Block name (PRE1/PRE2/POST)
3 Algorithm name
4 Preset memory or user memory name (first 11 characters)

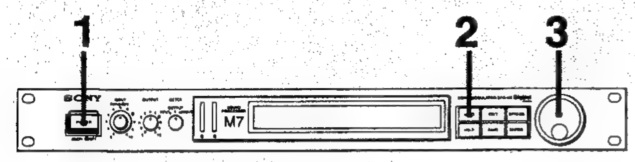
Block name will appear when the PRE-EF1 block load, PRE-EF2 block load or POST-EF block load is selected.

7. Turn the dial and select the memory number the specified block of which is to be copied.

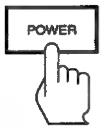
In the B. LOAD operation, the temporary buffer will be automatically loaded during editing only by turning the dial. The time from stopping the dial until loading can be set with "load time" of the system block. ("P" and "U" will stop blinking when loaded.)
The parameters of newly loaded blocks can be changed and comparative listening will also be possible.

Calling Up a Memory (LOAD)

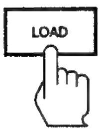
This operation calls up an effect stored in memory.

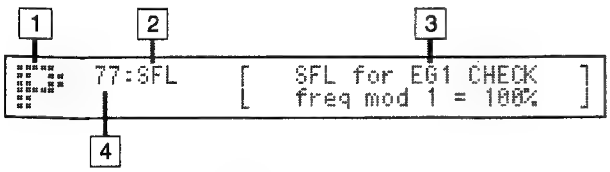


1. Turn on the power.



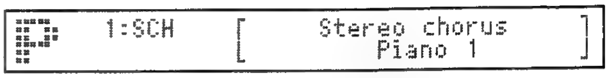
2. Press the LOAD button.





1 LOAD mode indication (P = Preset memory, U = User memory)
2 Algorithm name
3 Memory name
4 Memory number

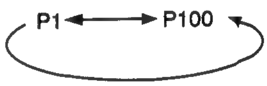
3. Turn the dial and select a desired memory number.



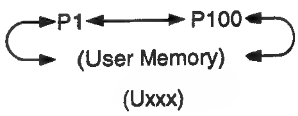
The effect of the selected memory number is automatically called up. When selecting "enter load" for "load form" in the system block, press the ENTER key after selecting the memory number. (If a number different from the effect currently called is selected, P or U will blink.)

Memory Numbers
A hundred settings are stored in the preset memory at time of shipment from the factory. These settings are displayed in endless order by turning the dial. If individually created settings are stored in the user memory, they will be inserted between P100 and P1 of the preset memory.

When shipped from factory



When stored in user memory



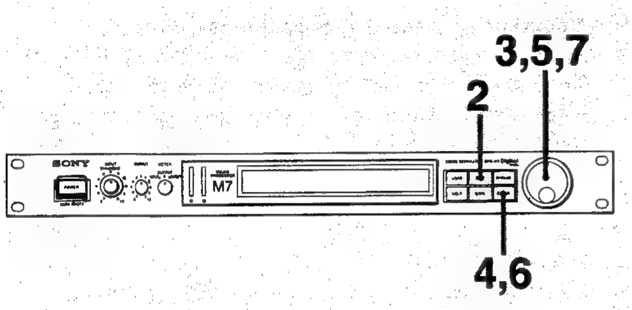
Algorithm Name
The "algorithm name" of LOAD mode indication screen shows the modulation block algorithm name.

Changing the Effect (EDIT)

To be continued ►

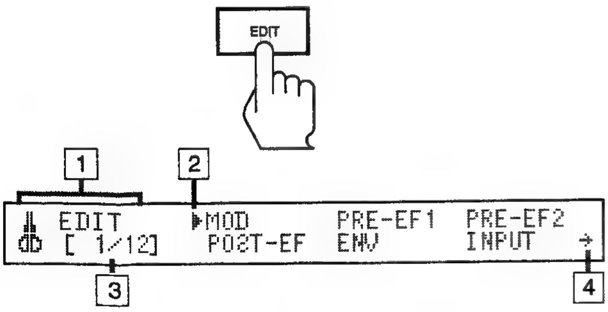
This function allows you to edit and create desired effects from those stored.

Example: Changing "effect level" of "Stereo Chorus (SCH)"



1. Call up a memory number to be changed (see page 60).

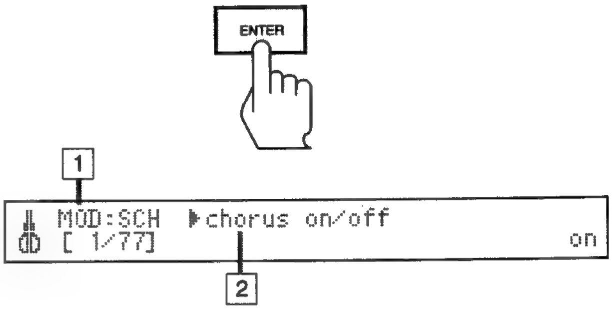
2. Press the EDIT button so that the block selecting screen will be displayed.



1 EDIT mode
2 Displayed on left side of the currently selected item.
3 This shows there are 12 selections and the first of these is selected.
4 This means there are more items following, If not, "↵" appears.

3. Turn the dial and select the block to be changed.

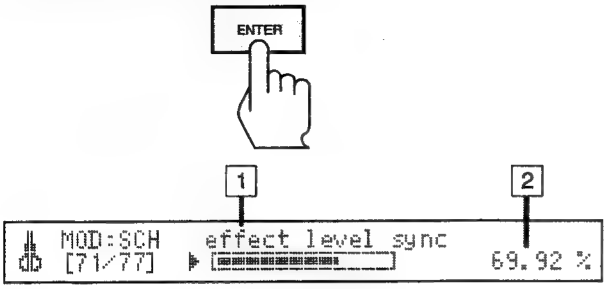
4. Press the ENTER button so that the parameter selecting screen will be displayed.



1 Block name and algorithm name
2 Parameter name

5. Turn the dial and select the parameter to be changed.

6. Press the ENTER button so that the parameter selecting screen will be displayed.



1 The bar graph changes according to the parameter value.
2 Parameter value

7. Turn the dial and change the parameter value.

To compare the result with the former effect
Each time the EDIT button is pressed while the parameter selecting screen is displayed, the range of comparative listening is changed in the order set with the "memory compare" parameter of the memory block in the parameter setting screen. By pressing the EDIT button several times, the former parameter value will resume. Sounds without any effect can be produced by pressing the BYPASS button even during comparative listening.

To change other parameters in the same block
1. After changing the parameter, press the ENTER button. The parameter selecting screen will be displayed.
2. Repeat steps **3** to **7** on the previous page to change other parameters as well.

To abort operation and restore the former memory setting
1. Press the LOAD button.
   Once the former effect resumes, all the parameters you have been setting are deleted, with the message "Parameters have been changed. Are you sure you want to load? Yes – ENTER No – EDIT".
   If you accept deletion of the parameters being changed, press the ENTER button. Otherwise, press the EDIT button to store the effect you have created by using the SAVE function. (See page 63.)
2. Press the ENTER button.
   The former memory resumes.

To enter the date and user information in the system block
Press the EDIT button and move the cursor.

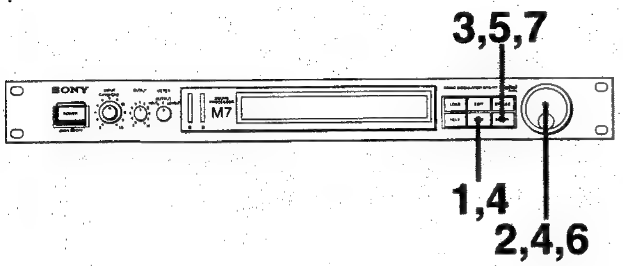
To change parameters of different blocks
1. Press the ENTER button after changing a parameter. The parameter selecting screen will be displayed.
2. Press the EDIT button (or press the ENTER button after selecting "QUIT" with the dial.
   The block selecting screen will be displayed.
3. Repeat steps 1 and 2 above to do the same for other parameters.

What is sync?
This is an indirect parameter that making different parameter values per channel the same. "sync" will be displayed after the parameter name. Even if different values are set in ch1 and ch2, the values will become the same if "sync" is executed. In other words, the value for ch1 is also applied for ch2.

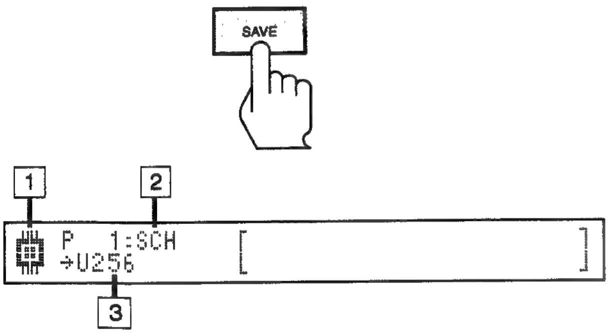
Changing units
Although units are generally changed in the system block, they can also be changed in the parameter selecting screen by pressing the ENTER button while pressing the HELP button.

Saving the changed effects (SAVE)

You can save the changed effects resulting from parameter values you have changed with the Edit function



1. Press the SAVE button.

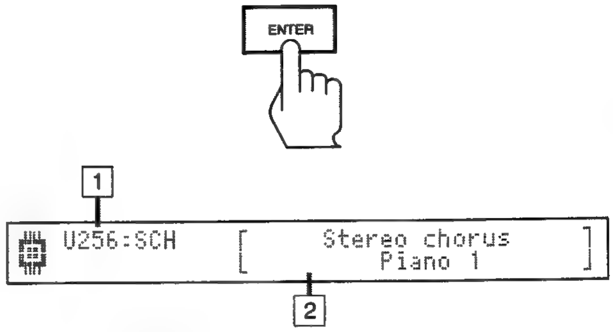


1 Save mode display
2 Algorithm name
3 User memory number

If you designate a memory number in which an effect is already stored, the algorithm name and memory name will be displayed after the user memory number.

2. Turn the dial and assign a number to the edited effect.

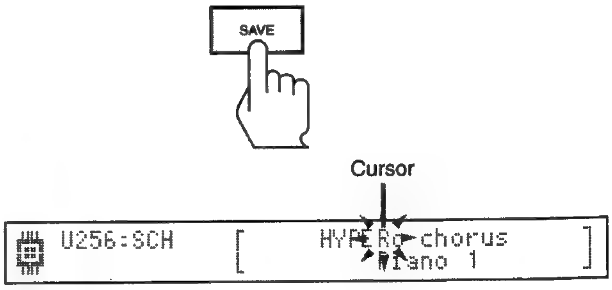
3. Press the ENTER button.



1 User memory number
2 Memory name of the original effect is displayed.

You cannot store the effect in the protected memory number indicated with " " unless you release the protection. (See page 56.)

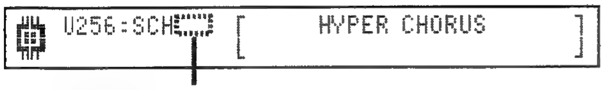
4. Denominate the memory.



Use the dial for selecting characters and the SAVE button for cursor movement. The cursor advances each time the SAVE button is pressed. The characters are aligned in the order of 0 - 9, A - Z, symbols and dot patterns. When desiring to delete the memory name of the original effect, move the cursor to the head of the memory name indication area. Select "all clear" with the dial and press the ENTER button. ("all clear" lies before the numbers.)

5. Press the ENTER button after denominating

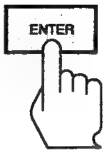
6. Turn the dial and set the memory protection if necessary.



If the memory protection is required, display the "🔒" symbol.

Turning the dial changes on/off for the "🔒" symbol.

7. Press the ENTER button



When the effect is saved, the "▣" changes to "U".

If there is not enough user memory remaining, the effect you have created may not be saved. You will see the message, "Memory full! need xxx byte more." In this case, you have to delete unnecessary memory (see the memory block on page 56) so that the memory area will have enough space to save the effect. If the effect is saved, "completed" is indicated and the SAVE mode is disengaged automatically to the LOAD mode.

What is memory protection?
Memory protection prevents the effect you have saved from being deleted by overwriting. The protected memory number cannot be used for overwriting unless you release the protection in the memory block.

MIDI Implementation Chart

To be continued ►

When transmitting data between DPS-M7 and other devices by using MIDI, sending and receiving is possible with the data format shown below and under the conditions of the implementation chart on page 68.

Channel voice message

* 1 1 0 0	n n n n	Program change & channel number(n n n n =0 – 15)
0 p p p	p p p p	Program number(p p p p p p p =0 – 127)
* 1 0 1 1	n n n n	Control change & channel number(n n n n =0 – 15)
0 c c c	c c c c	Control number(c c c c c c c =0 – 127)
0 v v v	v v v v	Control value(v v v v v v v =0 – 127)
* 1 0 0 1	n n n n	Note on & channel number(n n n n =0 – 15)
0 k k k	k k k k	Note number(k k k k k k k =0 – 127)
0 v v v	v v v v	Note on velocity(v v v v v v v =0 – 127)
* 1 1 0 1	n n n n	Channel pressure(after-touth) & channel number(n n n n =0 – 15)
0 v v v	v v v v	Pressure value(v v v v v v v =0 – 127)

Channel mode message

* 1 0 1 1	n n n n	Mode message & channel number(nnnn=0 – 15)	
0 c c c	c c c c	Omni mode off	Omni mode on
		ccccccc=124	ccccccc=125
0 v v v	v v v v	vvvvvvv=0	vvvvvvv=0

System exclusive message

* 1 1 1 1	0 0 0 0(F0)	Exclusive status	
0 1 0 0	1 1 0 0(4C)	SONY ID	Exclusive Header
0 0 0 0	n n n n(0n)	Global channel(nnnn=0 – 15)	Exclusive Header
0 0 0 1	0 0 1 1(13)	DPS-M7 ID	Exclusive Header
0 c c c	c c c c	Command	
0 d d d	d d d d	Data	
:		Data	
0 d d d	d d d d	Data	
1 1 1 1	0 1 1 1(F7)	End of Exclusive-EOX	

ALL DATA DUMP REQUEST (Receive)
Command : 0001 0000(10)
Data : None

ALL USER MEMORY DUMP REQUEST (Receive)
Command : 0001 0001(11)
Data : None

SYSTEM DUMP REQUEST (Receive)
Command : 0001 0010(12)
Data : None

MIDI DUMP REQUEST (Receive)
Command : 0001 0011(13)
Data : None

USER MEMORY DUMP REQUEST (Receive)
Command : 0001 010n (14or15)
bit 7
Data : 0 nnn nnnn   nnnnnnnn : User memory number-1(0 – 255)
bit 654 3210   bit 76543210

ALL DATA DUMP (Send/Receive) (ALL USER MEMORY + SYSTEM + MIDI)
Command : 0001 1000(18)
Data : 0ddd dddd ....   ddddddd : Data(see note 1,6)

ALL USER MEMORY DUMP (Send/Receive)
Command : 0001 1001(19)
Data : 0ddd dddd ....   ddddddd : Data(see note 1,2)

```
SYSTEM DUMP (Send/Receive)
  Command : 0001 1010 (1A)
  Data    : 0ddd dddd....              ddddddd : Data (see note 1,2)
MIDI DUMP (Send/Receive)
  Command : 0001 1011 (1B)
  Data    : 0ddd dddd....              ddddddd : Data (see note 1,4)
USER MEMORY DUMP (Send/Receive)
  Command : 0001 110n(1C or 1D)
                bit 7
  Data    : 0 n n n  n n n n
           bit6 5 4  3 2 1 0
          : 0ddd dddd..nnnnnnn           n n n n n n n n :User memory number-1 (0 – 255)
                                     bit 7 6 5 4 3 2 1 0
                                         ddddddd : Data (see note 1,5)

START ADDRESS TRANSFER (Receive)
  Command : 0010 0000(20)
  Data    : 0 a a a  a a a a
          bit 6 5 4  3 2 1 0
          : 0 a a a  a a a a
          bit D C B  A 9 8 7
          : 0 0 0 a  a a a a                a a a a a a a a a a a a a a a a a a a : Start address (0h – 7FFFFh)
          bit     12 1110F E           bit 121110F E D C B A 9 8 7 6 5 4 3 2 1 0

DATA TRANSFER (Receive)
  Command : 0100 0000(40)
  Data    : 0 a a a  a a a a
          bit 6 5 4  3 2 1 0
          : 0 a a a  a a a a
          bit D C B  A 9 8 7
          : 0 0 0 a  a a a a
          bit     12 1110F E                a a a a a a a a a a a a a a a a a a a : Start address (0h – 7FFFh)
          : 0ddd dddd........          bit 121110F E D C B A 9 8 7 6 5 4 3 2 1
                                           ddddddd : (see note1)
note1-dd:Data format
      0ddd dddd   0ddd dddd   0ddd dddd   0ddd dddd–
  bit 765  4321    076 5432    107 6543    210 7654
       ←    dd0    →←     dd1   →←   dd2     →← dd3
      0ddd dddd   0ddd dddd   0ddd dddd   0ddd dddd....
  bit 321  0765    432 1076    543 2107    654 3210
      dd3 →←      dd4    →←    dd5    →←   dd6     →←
note2-ALL USER MEMORY DUMP FORMAT
      dd0    ~ dd513           : USER MEMORY FAT
      dd514 ~ dd28113(max)  : USER MEMORY DATA
note3-SYSTEM DUMP FORMAT
      dd0    ~ dd47 : SYSTEM DATA
note4-MIDI DUMP FORMAT
      dd0    ~ dd257 : MIDI DATA
note5-USER MEMORY DUMP FORMAT
      dd0    ~ dd325(max) : USER MEMORY DATA
note6-ALL DATA DUMP FORMAT
      dd0    ~ dd47  : SYSTEM DATA
      dd48   ~ dd305 : MIDI DATA
      dd306 ~ dd819 : USER MEMORY FAT
      dd820 ~ dd28419 : USER MEMORY DATA
```

To be continued ►

Universal system exclusive message

INQUIRY MESSAGE

IDENTITY REQUEST (Receive)

* 1 1 1 1	0 0 0 0	(F 0)	Exclusive status	Universal System Exclusive Non-Real Time Header
0 1 1 1	1 1 1 0	(7 E)	Non realtime message	
0 0 0 0	n n n n	(0 n)	Global channel(nnnn=0 – 15)	
0 0 0 0	0 1 1 0	(0 6)	Inquiry message	
0 0 0 0	0 0 0 1	(0 1)	Identity request	
1 1 1 1	0 1 1 1	(F 7)	End of Exclusive – EOX	

IDENTITY REPLY (Send)

* 1 1 1 1	0 0 0 0	(F O)	Exclusive status	Universal System Exclusive Non-Real Time Header
0 1 1 1	1 1 1 0	(7 E)	Non realtime message	
0 0 0 0	n n n n	(0 n)	Global channel(nnnn=0 – 15)	
0 0 0 0	0 1 1 0	(0 6)	Inquiry message	
0 0 0 0	0 0 1 0	(0 2)	Identity reply	
0 1 0 0	1 1 0 0	(4 C)	SONY ID	
0 0 0 0	0 0 0 1	(0 1)		
0 0 0 0	0 0 0 0	(0 0)		
0 0 0 0	0 0 1 1	(0 3)	DPS-M7 ID	
0 0 0 0	0 0 0 0	(0 0)		
0 s s s	s s s s	(s s)	Software version	
0 s s s	s s s s	(s s)		
0 s s s	s s s s	(s s)		
0 s s s	s s s s	(s s)		
1 1 1 1	0 1 1 1	(F 7)	End of Exclusive – EOX	

DIGITAL SONIC MODULATOR DPS-M7
MIDI Implementation Chart

Date: 10 Dec. '91
Version: 1.0

Function ...		Transmitted	Recognized	Remarks
Basic Channel	Default Changed	× ×	1 – 16 1 – 16	Memorized
Mode	Default Messages Altered	× × **************	OMNI ON/OFF ×	Memorized
Note Number :	True voice	**************	○ 0 – 127	No sound
Velocity	Note ON Note OFF	× ×	○9n, V=0 – 127 ×	
After Touch	Key's Ch's	× ×	× ○	
Pitch Bend		×	×	
Control Change	0 – 120	×	○	
Prog Change :	True #	× **************	○0 – 127	
System Exclusive		○	○	
Common	: Song Pos : Song Sel : Tune	× × ×	× × ×	
System Real Time	: Clock : Commands	× ×	× ×	
Aux Messages	: Local ON/OFF : All Notes OFF : Active Sense : Reset	× × × ×	× × × ×	
Notes				

Model 1 : OMNI ON, POLY Model 2 : OMNI ON, MONO ○ : Yes
Model 3 : OMNI OFF, POLY Model 4 : OMNI OFF, MONO × : No

Classification Chart for Editing

The chart below lists all the blocks algorithm and parameters which you can edit.

EDIT

BLOCK	ALGORITHM	PARAMETER
MOD	Algorithm0:OFF	see page 70
	Algorithm1:SCH	see page 70
	Algorithm2:DCH	see page 70
	Algorithm3:MCH	see page 71
	Algorithm4:BCH	see page 71
	Algorithm5:SPS	see page 72
	Algorithm6:BPS	see page 72
	Algorithm7:PSM	see page 73
	Algorithm8:RVS	see page 73
	Algorithm9:ENS	see page 73
	Algorithm10:MPH	see page 74
	Algorithm11:SFL	see page 74
	Algorithm12:MFL	see page 75
	Algorithm13:MDL	see page 75
	Algorithm14:SPM	see page 76
	Algorithm15:SPA	see page 76
	Algorithm16:HPA	see page 76
	Algorithm17:DOP	see page 76
	Algorithm18:VIB	see page 76
	Algorithm19:RNG	see page 77
	Algorithm20:RTY	see page 77
PRE-EF1&2	Algorithm0:OFF	see page 77
	Algorithm1:SEQ	see page 77
	Algorithm2:SXE	see page 77
	Algorithm3:DEX	see page 77
	Algorithm4:GTE	see page 77
	Algorithm5:CMP	see page 77
POST-EF	Algorithm0:OFF	see page 78
	Algorithm1:SEQ	see page 78
	Algorithm2:SXE	see page 78
	Algorithm3:DEX	see page 78
	Algorithm4:GTE	see page 78
ENV	Algorithm0:OFF	see page 78
	Algorithm1:EVF	see page 78
	Algorithm2:EG1	see page 78
	Algorithm3:EG2	see page 78
INPUT		see page 78
OUTPUT		see page 78
LCL.MIDI		see page 78
SYSTEM		see page 78
MEMORY		see page 79
SYS.MIDI		see page 79
B.LOAD		see page 79

Classification Chart for Editing

BLOCK	ALGORITHM	PARAMETER

- EDIT
  - MOD
    - ALG0:OFF
      - QUIT
    - ALG1:SCH
      - chorus on/off
      - frequency scale
      - depth scale
      - time scale
      - LFO frequency sync/ch1/ch2
      - LFO freq modulation sync/ch1/ch2
      - LFO freq mod phase sync/ch1/ch2
      - LFO depth 1 sync/ch1/ch2
      - LFO depth 2 sync/ch1/ch2
      - LFO depth 1 modulation sync/ch1/ch2
      - LFO depth 1 mod phase sync/ch1/ch2
      - LFO depth 2 modulation sync/ch1/ch2
      - LFO depth 2 mod phase sync/ch1/ch2
      - LFO wave form sync/ch1/ch2
      - LFO phase sync/ch1/ch2
      - direct delay time sync/ch1/ch2
      - predelay 1 time sync/ch1/ch2
      - predelay 2 time sync/ch1/ch2
      - output 1 level sync/ch1/ ch2
      - output 1 phase sync/ch1/ ch2
      - panpot 1 level sync/ch1/ch2
      - output 2 level sync/ch1/ch2
      - output 2 phase sync/ch1/ch2
      - panpot 2 level sync/ch1/ch2
      - direct level sync/ch1/ch2
      - direct phase sync/ch1/ch2
      - effect level sync/ch1/ch2
      - effect phase sync/ch1/ch2
      - QUIT

1 2

(to be continued)

BLOCK	ALGORITHM	PARAMETER

1 2

- ALG2:DCH
  - chorus on/off
  - depth scale
  - time scale
  - LFO frequency
  - LFO freq modulation
  - LFO freq mod phase
  - LFO depth sync/ch1/ch2
  - LFO depth modulation sync/ch1/ch2
  - LFO depth mod phase sync/ch1/ch2
  - LFO wave form sync/ch1/ch2
  - LFO 1 phase sync/ch1/ch2
  - LFO 2 phase sync/ch1/ch2
  - LFO 3 phase sync/ch1/ch2
  - LFO 4 phase sync/ch1/ch2
  - LFO 5 phase sync/ch1/ch2
  - predelay 1 time sync/ch1/ch2
  - predelay 2 time sync/ch1/ch2
  - predelay 3 time sync/ch1/ch2
  - predelay 4 time sync/ch1/ch2
  - predelay 5 time sync/ch1/ch2
  - output 1 level sync/ch1/ch2
  - output 1 phase sync/ch1/ch2
  - panpot 1 level sync/ch1/ch2
  - output 2 level sync/ch1/ch2
  - output 2 phase sync/ch1/ch2
  - panpot 2 level sync/ch1/ch2
  - output 3 level sync/ch1/ch2
  - output 3 phase sync/ch1/ch2
  - panpot 3 level sync/ch1/ch2
  - output 4 level sync/ch1/ch2
  - output 4 phase sync/ch1/ch2
  - panpot 4 level sync/ch1/ch2
  - output 5 level sync/ch1/ch2
  - output 5 phase sync/ch1/ch2
  - panpot 5 level sync/ch1/ch2
  - direct level sync/ch1/ch2
  - direct phase sync/ch1/ch2
  - effect level sync/ch1/ch2
  - effect phase sync/ch1/ch2
  - effect modulation sync/ch1/ch2
  - effect mod phase sync/ch1/ch2
  - QUIT

3 4

(to be continued)

To be continued ▶

BLOCK	ALGORITHM	PARAMETER

3 4

- ALG3:MCH
  - chorus on/off
  - frequency scale
  - depth scale
  - time scale
  - chorus mode
  - LFO 1 frequency sync/ch1/ch2
  - LFO 2 frequency sync/ch1/ch2
  - LFO 3 frequency sync/ch1/ch2
  - LFO 1 freq modulation sync/ch1/ch2
  - LFO 1 freq mod phase sync/ch1/ch2
  - LFO 2 freq modulation sync/ch1/ch2
  - LFO 2 freq mod phase sync/ch1/ch2
  - LFO 3 freq modulation sync/ch1/ch2
  - LFO 3 freq mod phase sync/ch1/ch2
  - LFO 1 depth sync/ch1/ch2
  - LFO 2 depth sync/ch1/ch2
  - LFO 3 depth sync/ch1/ch2
  - LFO 1 depth modulation sync/ch1/ch2
  - LFO 1 depth mod phase sync/ch1/ch2
  - LFO 2 depth modulation sync/ch1/ch2
  - LFO 2 depth mod phase sync/ch1/ch2
  - LFO 3 depth modulation sync/ch1/ch2
  - LFO 3 depth mod phase sync/ch1/ch2
  - LFO 1 wave form sync/ch1/ch2
  - LFO 2 wave form sync/ch1/ch2
  - LFO 3 wave form sync/ch1/ch2
  - LFO 1 phase sync/ch1/ch2
  - LFO 2 phase sync/ch1/ch2
  - LFO 3 phase sync/ch1/ch2
  - direct delay time sync/ch1/ch2
  - predelay 1 time sync/ch1/ch2
  - feedback 1 level sync/ch1/ch2
  - feedback 1 phase sync/ch1/ch2
  - hi freq fdbk damping 1 sync/ch1/ch2
  - predelay 2 time sync/ch1/ch2
  - feedback 2 level sync/ch1/ch2
  - feedback 2 phase sync/ch1/ch2
  - hi freq fdbk damping 2 sync/ch1/ch2
  - predelay 3 time sync/ch1/ch2
  - feedback 3 level sync/ch1/ch2
  - feedback 3 phase sync/ch1/ch2
  - hi freq fdbk damping 3 sync/ch1/ch2
  - output 1 level sync/ch1/ch2
  - output 1 phase sync/ch1/ch2
  - panpot 1 level sync/ch1/ch2
  - output 2 level sync/ch1/ch2
  - output 2 phase sync/ch1/ch2
  - panpot 2 level sync/ch1/ch2
  - output 3 level sync/ch1/ch2
  - output 3 phase sync/ch1/ch2
  - panpot 3 level sync/ch1/ch2
  - direct level sync/ch1/ch2
  - direct phase sync/ch1/ch2
  - effect level sync/ch1/ch2
  - effect phase sync/ch1/ch2
  - effect modulation sync/ch1/ch2
  - effect mod phase sync/ch1/ch2
  - QUIT

5 6

BLOCK	ALGORITHM	PARAMETER

5 6

- ALG4:BCH
  - chorus on/off
  - frequency scale
  - depth scale
  - time scale
  - treble frequency sync/ch1/ch2
  - treble level sync/ch1/ch2
  - PEQ frequency sync/ch1/ch2
  - PEQ level sync/ch1/ch2
  - PEQ q sync/ch1/ch2
  - bass frequency sync/ch1/ch2
  - bass level sync/ch1/ch2
  - LFO 1 frequency sync/ch1/ch2
  - LFO 2 frequency sync/ch1/ch2
  - LFO 3 frequency sync/ch1/ch2
  - LFO 1 freq modulation sync/ch1/ch2
  - LFO 1 freq mod phase sync/ch1/ch2
  - LFO 2 freq modulation sync/ch1/ch2
  - LFO 2 freq mod phase sync/ch1/ch2
  - LFO 3 freq modulation sync/ch1/ch2
  - LFO 3 freq mod phase sync/ch1/ch2
  - LFO 1 depth sync/ch1/ch2
  - LFO 2 depth sync/ch1/ch2
  - LFO 3 depth sync/ch1/ch2
  - LFO 1 depth modulation sync/ch1/ch2
  - LFO 1 depth mod phase sync/ch1/ch2
  - LFO 2 depth modulation sync/ch1/ch2
  - LFO 2 depth mod phase sync/ch1/ch2
  - LFO 3 depth modulation sync/ch1/ch2
  - LFO 3 depth mod phase sync/ch1/ch2
  - LFO 1 wave form sync/ch1/ch2
  - LFO 2 wave form sync/ch1/ch2
  - LFO 3 wave form sync/ch1/ch2
  - LFO 1 phase sync/ch1/ch2
  - LFO 2 phase sync/ch1/ch2
  - LFO 3 phase sync/ch1/ch2
  - direct delay time sync/ch1/ch2
  - predelay 1 time sync/ch1/ch2
  - predelay 2 time sync/ch1/ch2
  - predelay 3 time sync/ch1/ch2
  - output 1 level sync/ch1/ch2
  - output 1 phase sync/ch1/ch2
  - panpot 1 level sync/ch1/ch2
  - output 2 level sync/ch1/ch2
  - output 2 phase sync/ch1/ch2
  - panpot 2 level sync/ch1/ch2
  - output 3 level sync/ch1/ch2
  - output 3 phase sync/ch1/ch2
  - panpot 3 level sync/ch1/ch2
  - direct level sync/ch1/ch2
  - direct phase sync/ch1/ch2
  - effect level sync/ch1/ch2
  - effect phase sync/ch1/ch2
  - effect modulation sync/ch1/ch2
  - effect mod phase sync/ch1/ch2
  - QUIT

7 8

Classification Chart for Editing

BLOCK	ALGORITHM	PARAMETER

7 ▽ — ▽ 9

8 ▽

- ALG5:SPS
  - (pitch shifter mode = fixed)
    - pitch shifter on/off
    - time scale
    - pitch shifter mode
    - pitch sync/ch1/ch2
    - pitch modulation sync/ch1/ch2
    - pitch mod phase sync/ch1/ch2
    - direct delay time sync/ch1/ch2
    - predelay time sync/ch1/ch2
    - feedback level sync/ch1/ch2
    - feedback phase sync/ch1/ch2
    - direct level sync/ch1/ch2
    - direct phase sync/ch1/ch2
    - effect level sync/ch1/ch2
    - effect phase sync/ch1/ch2
    - QUIT
  - (pitch shifter mode = harmony)
    - pitch shifter on/off
    - time scale
    - pitch shifter mode
    - scale type
    - base key
    - note interval sync/ch1/ch2
    - note interval C sync/ch1/ch2
    - :
    - note interval B sync/ch1/ch2
    - pitch modulation sync/ch1/ch2
    - pitch mod phase sync/ch1/ch2
    - direct delay time sync/ch1/ch2
    - predelay time sync/ch1/ch2
    - feedback level sync/ch1/ch2
    - feedback phase sync/ch1/ch2
    - direct level sync/ch1/ch2
    - direct phase sync/ch1/ch2
    - effect level sync/ch1/ch2
    - effect phase sync/ch1/ch2
    - QUIT
- ALG6:BPS
  - (pitch shifter mode = fixed)
    - pitch shifter on/off
    - time scale
    - pitch shifter mode
    - treble frequency snyc/ch1/ch2
    - treble level sync/ch1/ch2
    - bass frequency sync/ch1/ch2
    - bass level sync/ch1/ch2
    - pitch 1 sync/ch1/ch2
    - pitch 2 sync/ch1/ch2
    - pitch modulation 1 sync/ch1/ch2
    - pitch mod 1 phase sync/ch1/ch2
    - pitch modulation 2 sync/ch1/ch2
    - pitch mod 2 phase sync/ch1/ch2
    - direct delay time sync/ch1/ch2
    - predelay 1 time sync/ch1/ch2
    - feedback 1 level sync/ch1/ch2
    - feedback 1 phase sync/ch1/ch2
    - predelay 2 time sync/ch1/ch2
    - feedback 2 level sync/ch1/ch2
    - feedback 2 phase sync/ch1/ch2
    - output 1 level sync/ch1/ch2
    - output 1 phase sync/ch1/ch2
    - output 2 level sync/ch1/ch2
    - output 2 phase sync/ch1/ch2
    - direct level sync/ch1/ch2
    - direct phase sync/ch1/ch2
    - effect level sync/ch1/ch2
    - effect phase sync/ch1/ch2
    - QUIT

▽ 10 ▽ 11

BLOCK	ALGORITHM	PARAMETER

9 ▽ — ▽ 12

10 ▽ — ▽ 13

11 ▽

- (pitch shifter mode = fixed)
  - pitch shifter on/off
  - time scale
  - pitch shifter mode
  - treble frequency sync/ch1/ch2
  - treble level sync/ch1/ch2
  - bass frequency sync/ch1/ch2
  - bass level sync/ch1/ch2
  - scale type
  - base key
  - note interval 1 sync/ch1/ch2
  - note interval 2 sync/ch1/ch2
  - note interval 1: C sync/ch1/ch2
  - :
  - note interval 1: B sync/ch1/ch2
  - note interval 2: C sync/ch1/ch2
  - :
  - note interval 2: B sync/ch1/ch2
  - pitch modulation 1 sync/ch1/ch2
  - pitch mod 1 phase sync/ch1/ch2
  - pitch modulation 2 sync/ch1/ch2
  - pitch mod 2 phase sync/ch1/ch2
  - direct delay time sync/ch1/ch2
  - predelay 1 time sync/ch1/ch2
  - feedback 1 level sync/ch1/ch2
  - feedback 1 phase sync/ch1/ch2
  - predelay 2 time sync/ch1/ch2
  - feedback 2 level sync/ch1/ch2
  - feedback 2 phase sync/ch1/ch2
  - output 1 level sync/ch1/ch2
  - output 1 phase sync/ch1/ch2
  - output 2 level sync/ch1/ch2
  - output 2 phase sync/ch1/ch2
  - direct level sync/ch1/ch2
  - direct phase sync/ch1/ch2
  - effect level sync/ch1/ch2
  - effect phase sync/ch1/ch2
  - QUIT

To be continued ▶

BLOCK	ALGORITHM	PARAMETER

12 13

- ALG7:PSM
  - (pitch shifter mode = fixed)
    - pitch shift mod on/off
    - frequency scale
    - depth scale
    - time scale
    - pitch shifter mode
    - pitch sync/ch1/ch2
    - pitch modulation sync/ch1/ch2
    - pitch mod phase sync/ch1/ch2
    - predelay time sync/ch1/ch2
    - feedback level sync/ch1/ch2
    - feedback phase sync/ch1/ch2
    - hi freq fdbk damping sync/ch1/ch2
    - LFO frequency sync/ch1/ch2
    - LFO freq modulation sync/ch1/ch2
    - LFO freq mod phase sync/ch1/ch2
    - LFO depth sync/ch1/ch2
    - LFO depth modulation sync/ch1/ch2
    - LFO depth mod phase sync/ch1/ch2
    - LFO wave form sync/ch1/ch2
    - LFO phase sync/ch1/ch2
    - chorus delay time sync/ch1/ch2
    - chorus feedback level sync/ch1/ch2
    - chorus feedback phase sync/ch1/ch2
    - DRY : EFF mix balance sync/ch1/ch2
    - output level sync/ch1/ch2
    - output phase sync/ch1/ch2
    - panpot level sync/ch1/ch2
    - direct level sync/ch1/ch2
    - direct phase sync/ch1/ch2
    - effect level sync/ch1/ch2
    - effect phase sync/ch1/ch2
    - QUIT
  - (pitch shifter mode = harmony)
    - pitch shift mod on/off
    - frequency scale
    - depth scale
    - time scale
    - pitch shifter mode
    - scale type
    - base key
    - note interval sync/ch1/ch2
    - note interval C sync/ch1/ch2
    - :
    - note interval B sync/ch1/ch2
    - pitch modulation sync/ch1/ch2
    - pitch mod phase sync/ch1/ch2
    - predelay time sync/ch1/ch2
    - feedback level sync/ch1/ch2
    - feedback phase sync/ch1/ch2
    - hi freq fdbk damping sync/ch1/ch2
    - LFO frequency sync/ch1/ch2
    - LFO freq modulation sync/ch1/ch2
    - LFO freq mod phase sync/ch1/ch2
    - LFO depth sync/ch1/ch2
    - LFO depth modulation sync/ch1/ch2
    - LFO depth mod phase sync/ch1/ch2
    - LFO wave form sync/ch1/ch2
    - LFO phase sync/ch1/ch2
    - chorus delay time sync/ch1/ch2
    - chorus feedback level sync/ch1/ch2
    - chorus feedback phase sync/ch1/ch2
    - DRY : EFF mix balance sync/ch1/ch2
    - output level sync/ch1/ch2
    - output phase sync/ch1/ch2
    - panpot level sync/ch1/ch2
    - direct level sync/ch1/ch2
    - direct phase sync/ch1/ch2
    - effect level sync/ch1/ch2
    - effect phase sync/ch1/ch2
    - QUIT

14 15

(to be continued)

BLOCK	ALGORITHM	PARAMETER

14 15

- ALG8:RVS
  - reverse shift on/off
  - pitch sync/ch1/ch2
  - length sync/ch1/ch2
  - feedback level sync/ch1/ch2
  - feedback phase sync/ch1/ch2
  - hi freq fdbk damping sync/ch1/ch2
  - output level sync/ch1/ch2
  - output phase sync/ch1/ch2
  - panpot level sync/ch1/ch2
  - direct level sync/ch1/ch2
  - direct phase sync/ch1/ch2
  - effect level sync/ch1/ch2
  - effect phase sync/ch1/ch2
  - effect modulation sync/ch1/ch2
  - effect mod phase sync/ch1/ch2
  - QUIT
- ALG9:ENS
  - ensemble on/off
  - frequency scale
  - depth scale
  - time scale
  - LFO 1 frequency sync/ch1/ch2
  - LFO 2 frequency sync/ch1/ch2
  - LFO 1 freq modulation sync/ch1/ch2
  - LFO 1 freq mod phase sync/ch1/ch2
  - LFO 2 freq modulation sync/ch1/ch2
  - LFO 2 freq mod phase sync/ch1/ch2
  - LFO 1 depth sync/ch1/ch2
  - LFO 2 depth sync/ch1/ch2
  - LFO 1 depth modulation sync/ch1/ch2
  - LFO 1 depth mod phase sync/ch1/ch2
  - LFO 2 depth modulation sync/ch1/ch2
  - LFO 2 depth mod phase sync/ch1/ch2
  - LFO 1 wave form sync/ch1/ch2
  - LFO 2 wave form sync/ch1/ch2
  - LFO 1 phase 1 sync/ch1/ch2
  - LFO 1 phase 2 sync/ch1/ch2
  - LFO 1 phase 3 sync/ch1/ch2
  - LFO 2 phase 1 sync/ch1/ch2
  - LFO 2 phase 2 sync/ch1/ch2
  - LFO 2 phase 3 sync/ch1/ch2
  - direct delay time sync/ch1/ch2
  - predelay 1 time sync/ch1/ch2
  - predelay 2 time sync/ch1/ch2
  - predelay 3 time sync/ch1/ch2
  - output 1 level sync/ch1/ch2
  - output 1 phase sync/ch1/ch2
  - panpot 1level sync/ch1/ch2
  - output 2 level sync/ch1/ch2
  - output 2 phase sync/ch1/ch2
  - panpot 2 level sync/ch1/ch2
  - output 3 level sync/ch1/ch2
  - output 3 phase sync/ch1/ch2
  - panpot 3 level sync/ch1/ch2
  - direct level sync/ch1/ch2
  - direct phase sync/ch1/ch2
  - effect level sync/ch1/ch2
  - effect phase sync/ch1/ch2
  - effect modulation sync/ch1/ch2
  - effect mod phase sync/ch1/ch2
  - QUIT

16 17

(to be continued)

BLOCK	ALGORITHM	PARAMETER

16 17

ALG10:MPH
- phase shifter on/off
- frequency scale
- depth scale
- time scale
- stage number 1 sync/ch1/ch2
- stage number 2 sync/ch1/ch2
- mode sync/ch1/ch2
- LFO 1 frequency sync/ch1/ch2
- LFO 2 frequency sync/ch1/ch2
- LFO 1 freq modulation sync/ch1/ch2
- LFO 1 freq mod phase sync/ch1/ch2
- LFO 2 freq modulation sync/ch1/ch2
- LFO 2 freq mod phase sync/ch1/ch2
- LFO 1 depth sync/ch1/ch2
- LFO 2 depth sync/ch1/ch2
- manual 1 sync/ch1/ch2
- manual 2 sync/ch1/ch2
- manual 1 modulation sync/ch1/ch2
- manual 1 mod phase sync/ch1/ch2
- manual 2 modulation sync/ch1/ch2
- manual 2 mod phase sync/ch1/ch2
- S/H step frequency 1 sync/ch1/ch2
- S/H step frequency 2 sync/ch1/ch2
- S/H step freq mod 1 sync/ch1/ch2
- S/H step f. mod 1 phase sync/ch1/ch2
- S/H step freq mod 2 sync/ch1/ch2
- S/H step f. mod 2 phase sync/ch1/ch2
- LFO 1 wave form sync/ch1/ch2
- LFO 2 wave form sync/ch1/ch2
- LFO 1 phase sync/ch1/ch2
- LFO 2 phase sync/ch1/ch2
- resonance 1 level sync/ch1/ch2
- resonance 1 phase sync/ch1/ch2
- resonance 2 level sync/ch1/ch2
- resonance 2 phase sync/ch1/ch2
- direct delay time sync/ch1/ch2
- postdelay 1 time sync/ch1/ch2
- feedback 1 level sync/ch1/ch2
- feedback 1 phase sync/ch1/ch2
- postdelay 2 time sync/ch1/ch2
- feedback 2 level sync/ch1/ch2
- feedback 2 phase sync/ch1/ch2
- output 1 level sync/ch1/ch2
- output 1 phase sync/ch1/ch2
- panpot 1 level sync/ch1/ch2
- output 2 level sync/ch1/ch2
- output 2 phase sync/ch1/ch2
- panpot 2 level sync/ch1/ch2
- direct level sync/ch1/ch2
- direct phase sync/ch1/ch2
- effect level sync/ch1/ch2
- effect phase sync/ch1/ch2
- effect modulation sync/ch1/ch2
- effect mod phase sync/ch1/ch2
- QUIT

18 19

(to be continued)

BLOCK	ALGORITHM	PARAMETER

18 19

ALG11:SFL
- flanger on/off
- frequency scale
- depth scale
- time scale
- LFO 1 frequency sync/ch1/ch2
- LFO 2 frequency sync/ch1/ch2
- LFO 1 freq modulation sync/ch1/ch2
- LFO 1 freq mod phase sync/ch1/ch2
- LFO 2 freq modulation sync/ch1/ch2
- LFO 2 freq mod phase sync/ch1/ch2
- LFO 1 depth sync/ch1/ch2
- LFO 2 depth sync/ch1/ch2
- LFO 1 depth modulation sync/ch1/ch2
- LFO 1 depth mod phase sync/ch1/ch2
- LFO 2 depth modulation sync/ch1/ch2
- LFO 2 depth mod phase sync/ch1/ch2
- manual sync/ch1/ch2
- manual modulation sync/ch1/ch2
- manual mod phase sync/ch1/ch2
- LFO 1 wave form sync/ch1/ch2
- LFO 2 wave form sync/ch1/ch2
- LFO 1 phase sync/ch1/ch2
- LFO 2 phase sync/ch1/ch2
- phase shifter on/off sync/ch1/ch2
- shifting point sync/ch1/ch2
- shifting point phase sync/ch1/ch2
- direct delay time sync/ch1/ch2
- predelay time sync/ch1/ch2
- feedback level sync/ch1/ch2
- feedback phase sync/ch1/ch2
- feedback mode sync/ch1/ch2
- direct level sync/ch1/ch2
- direct phase sync/ch1/ch2
- effect level sync/ch1/ch2
- effect phase sync/ch1/ch2
- effect output mode sync/ch1/ch2
- QUIT

20 21

(to be continued)

To be continued ▶

BLOCK	ALGORITHM	PARAMETER

20 21

- ALG12:MFL
  - flanger on/off
  - frequency scale
  - depth scale
  - time scale
  - flanger mode
  - LFO 1 frequency sync/ch1/ch2
  - LFO 2 frequency sync/ch1/ch2
  - LFO 1 freq modulation sync/ch1/ch2
  - LFO 1 freq mod phase sync/ch1/ch2
  - LFO 2 freq modulation sync/ch1/ch2
  - LFO 2 freq mod phase sync/ch1/ch2
  - LFO 1 depth sync/ch1/ch2
  - LFO 2 depth sync/ch1/ch2
  - LFO 1 depth modulation sync/ch1/ch2
  - LFO 1 depth mod phase sync/ch1/ch2
  - LFO 2 depth modulation sync/ch1/ch2
  - LFO 2 depth mod phase sync/ch1/ch2
  - manual 1 sync/ch1/ch2
  - manual 2 sync/ch1/ch2
  - LFO 1 wave form sync/ch1/ch2
  - LFO 2 wave form sync/ch1/ch2
  - LFO 1 phase sync/ch1/ch2
  - LFO 2 phase sync/ch1/ch2
  - direct delay time sync/ch1/ch2
  - predelay 1 time sync/ch1/ch2
  - predelay 1 fdbk level sync/ch1/ch2
  - predelay 1 fdbk phase sync/ch1/ch2
  - hi freq fdbk damping 1 sync/ch1/ch2
  - predelay 2 time sync/ch1/ch2
  - predelay 2 fdbk level sync/ch1/ch2
  - predelay 2 fdbk phase sync/ch1/ch2
  - hi freq fdbk damping 2 sync/ch1/ch2
  - flanger 1 fdbk level sync/ch1/ch2
  - flanger 1 fdbk phase sync/ch1/ch2
  - flanger 2 fdbk level sync/ch1/ch2
  - flanger 2 fdbk phase sync/ch1/ch2
  - DRY : EFF mix balance 1 sync/ch1/ch2
  - DRY : EFF mix balance 2 sync/ch1/ch2
  - output 1 level sync/ch1/ch2
  - output 1 phase sync/ch1/ch2
  - output 2 level sync/ch1/ch2
  - output 2 phase sync/ch1/ch2
  - autopan 1 frequency sync/ch1/ch2
  - autopan 2 frequency sync/ch1/ch2
  - autopan 1 freq mod sync/ch1/ch2
  - autopan 1 f. mod phase sync/ch1/ch2
  - autopan 2 freq mod sync/ch1/ch2
  - autopan 2 f. mod phase sync/ch1/ch2
  - autopan 1 depth sync/ch1/ch2
  - autopan 2 depth sync/ch1/ch2
  - autopan 1 wave form sync/ch1/ch2
  - autopan 2 wave form sync/ch1/ch2
  - autopan 1 phase sync/ch1/ch2
  - autopan 2 phase sync/ch1/ch2
  - direct level sync/ch1/ch2
  - direct phase sync/ch1/ch2
  - effect level sync/ch1/ch2
  - effect phase sync/ch1/ch2
  - QUIT

22 23

BLOCK	ALGORITHM	PARAMETER

22 23

- ALG13:MDL
  - modulation delay on/off
  - frequency scale
  - depth scale
  - time scale
  - LFO frequency sync/ch1/ch2
  - LFO freq modulation sync/ch1/ch2
  - LFO freq mod phase sync/ch1/ch2
  - LFO depth sync/ch1/ch2
  - LFO depth modulation sync/ch1/ch2
  - LFO depth mod phase sync/ch1/ch2
  - LFO wave form sync/ch1/ch2
  - LFO phase sync/ch1/ch2
  - direct delay time sync/ch1/ch2
  - predelay time sync/ch1/ch2
  - main delay time sync/ch1/ch2
  - feedback level sync/ch1/ch2
  - feedback phase sync/ch1/ch2
  - main delay tap level sync/ch1/ch2
  - main delay tap phase sync/ch1/ch2
  - main delay tap panpot sync/ch1/ch2
  - sub delay tap 1 time sync/ch1/ch2
  - sub delay tap 1 level sync/ch1/ch2
  - sub delay tap 1 phase sync/ch1/ch2
  - sub delay tap 1 panpot sync/ch1/ch2
  - sub delay tap 2 time sync/ch1/ch2
  - sub delay tap 2 level sync/ch1/ch2
  - sub delay tap 2 phase sync/ch1/ch2
  - sub delay tap 2 panpot sync/ch1/ch2
  - sub delay tap 3 time sync/ch1/ch2
  - sub delay tap 3 level sync/ch1/ch2
  - sub delay tap 3 phase sync/ch1/ch2
  - sub delay tap 3 panpot sync/ch1/ch2
  - bass frequency sync/ch1/ch2
  - bass level sync/ch1/ch2
  - treble frequency sync/ch1/ch2
  - treble level sync/ch1/ch2
  - direct level sync/ch1/ch2
  - direct phase sync/ch1/ch2
  - effect level sync/ch1/ch2
  - effect phase sync/ch1/ch2
  - effect modulation sync/ch1/ch2
  - effect mod phase sync/ch1/ch2
  - QUIT

24 25

BLOCK	ALGORITHM	PARAMETER

24 25

- ALG14:SPM
  - spiral modulation on/off
  - frequency scale
  - depth scale
  - time scale
  - pitch mode sync/ch1/ch2
  - pitch LFO frequency sync/ch1/ch2
  - pitch LFO freq mod sync/ch1/ch2
  - pitch LFO freq mod phase sync/ch1/ch2
  - pitch LFO depth sync/ch1/ch2
  - pitch LFO phase 1 sync/ch1/ch2
  - pitch LFO phase 2 sync/ch1/ch2
  - predelay 1 time sync/ch1/ch2
  - feedback 1 level sync/ch1/ch2
  - feedback 1 phase sync/ch1/ch2
  - output 1 level sync/ch1/ch2
  - output 1 phase sync/ch1/ch2
  - predelay 2 time sync/ch1/ch2
  - feedback 2 level sync/ch1/ch2
  - feedback 2 phase sync/ch1/ch2
  - output 2 level sync/ch1/ch2
  - output 2 phase sync/ch1/ch2
  - autopan mode sync/ch1/ch2
  - autopan frequency sync/ch1/ch2
  - autopan freq mod sync/ch1/ch2
  - autopan freq mod phase sync/ch1/ch2
  - autopan depth sync/ch1/ch2
  - autopan phase 1 sync/ch1/ch2
  - autopan phase 2 sync/ch1/ch2
  - direct level sync/ch1/ch2
  - direct phase sync/ch1/ch2
  - effect level sync/ch1/ch2
  - effect phase sync/ch1/ch2
  - QUIT
- ALG15:SPA
  - stereo panner on/off
  - frequency scale
  - mode
  - manual limit min sync/ch1/ch2
  - manual limit max sync/ch1/ch2
  - LFO 1 frequency sync/ch1/ch2
  - LFO 2 frequency sync/ch1/ch2
  - LFO 1 freq modulation sync/ch1/ch2
  - LFO 1 freq mod phase sync/ch1/ch2
  - LFO 2 freq modulation sync/ch1/ch2
  - LFO 2 freq mod phase sync/ch1/ch2
  - LFO 1 wave form sync/ch1/ch2
  - LFO 2 wave form sync/ch1/ch2
  - LFO 1: LFO2 mix balance sync/ch1/ch2
  - LFO 1 start point sync/ch1/ch2
  - LFO 1 step sync/ch1/ch2
  - LFO 2 start point sync/ch1/ch2
  - LFO 2 step sync/ch1/ch2
  - trigger 1 select sync/ch1/ch2
  - trigger 1 threshold sync/ch1/ch2
  - trigger 2 select sync/ch1/ch2
  - trigger 2 threshold sync/ch1/ch2
  - QUIT

26 27

BLOCK	ALGORITHM	PARAMETER

26 27

- ALG16:HPA
  - haas panner on/off
  - frequency scale
  - mode
  - manual limit min sync/ch1/ch2
  - manual limit max sync/ch1/ch2
  - LFO 1 frequency sync/ch1/ch2
  - LFO 2 frequency sync/ch1/ch2
  - LFO 1 freq modulation sync/ch1/ch2
  - LFO 1 freq mod phase sync/ch1/ch2
  - LFO 2 freq modulation sync/ch1/ch2
  - LFO 2 freq mod phase sync/ch1/ch2
  - LFO 1 wave form sync/ch1/ch2
  - LFO 2 wave form sync/ch1/ch2
  - LFO 1: LFO 2 mix balance sync/ch1/ch2
  - LFO 1 start point sync/ch1/ch2
  - LFO 1 step sync/ch1/ch2
  - LFO 2 start point sync/ch1/ch2
  - LFO 2 step sync/ch1/ch2
  - trigger 1 select sync/ch1/ch2
  - trigger 1 threshold sync/ch1/ch2
  - trigger 2 select sync/ch1/ch2
  - trigger 2 threshold sync/ch1/ch2
  - QUIT
- ALG17:DOP
  - doppler on/off
  - maximum distance sync/ch1/ch2
  - minimum distance sync/ch1/ch2
  - speed sync/ch1/ch2
  - speed modulation sync/ch1/ch2
  - speed mod phase sync/ch1/ch2
  - direction sync/ch1/ch2
  - trigger select sync/ch1/ch2
  - trigger threshold sync/ch1/ch2
  - ambience sync/ch1/ch2
  - output level sync/ch1/ch2
  - output phase sync/ch1/ch2
  - QUIT
- ALG18:VIB
  - vibrato + tremolo on/off
  - frequency scale
  - depth scale
  - time scale
  - LFO 1 frequency sync/ch1/ch2
  - LFO 2 frequency sync/ch1/ch2
  - LFO 1 freq modulation sync/ch1/ch2
  - LFO 1 freq mod phase sync/ch1/ch2
  - LFO 2 freq modulation sync/ch1/ch2
  - LFO 2 freq mod phase sync/ch1/ch2
  - LFO 1 depth sync/ch1/ch2
  - LFO 2 depth sync/ch1/ch2
  - LFO 1 depth modulation sync/ch1/ch2
  - LFO 1 depth mod phase sync/ch1/ch2
  - LFO 2 depth modulation sync/ch1/ch2
  - LFO 2 depth mod phase sync/ch1/ch2
  - LFO 1 wave form sync/ch1/ch2
  - LFO 2 wave form sync/ch1/ch2
  - LFO 1 phase sync/ch1/ch2
  - LFO 2 phase sync/ch1/ch2
  - trigger select sync/ch1/ch2
  - trigger threshold sync/ch1/ch2
  - trigger delay sync/ch1/ch2
  - rising time sync/ch1/ch2
  - falling time sync/ch1/ch2
  - tremolo depth sync/ch1/ch2
  - tremolo LFO phase sync/ch1/ch2
  - predelay time sync/ch1/ch2
  - output level sync/ch1/ch2
  - output phase sync/ch1/ch2
  - QUIT

28 29

To be continued ►

BLOCK	ALGORITHM	PARAMETER

28 ▽ → 30 ▽

29 ▽

- ALG19:RNG
  - ring modulator on/off
  - frequency scale
  - time scale
  - OSC frequecy sync/ch1/ch2
  - OSC freq modulation sync/ch1/ch2
  - OSC freq mod phase sync/ch1/ch2
  - OSC phase sync/ch1/ch2
  - ring modulator mode sync/ch1/ch2
  - direct delay time sync/ch1/ch2
  - main delay time sync/ch1/ch2
  - feedback level sync/ch1/ch2
  - feedback phase sync/ch1/ch2
  - main delay tap 1 time sync/ch1/ch2
  - main delay tap 1 level sync/ch1/ch2
  - main delay tap 1 phase sync/ch1/ch2
  - main delay tap 1 pan sync/ch1/ch2
  - main delay tap 2 time sync/ch1/ch2
  - main delay tap 2 level sync/ch1/ch2
  - main delay tap 2 phase sync/ch1/ch2
  - main delay tap 2 pan sync/ch1/ch2
  - main delay tap 3 time sync/ch1/ch2
  - main delay tap 3 level sync/ch1/ch2
  - main delay tap 3 phase sync/ch1/ch2
  - main delay tap 3 pan sync/ch1/ch2
  - bass frequency sync/ch1/ch2
  - bass level sync/ch1/ch2
  - treble frequecy sync/ch1/ch2
  - treble level sync/ch1/ch2
  - LPF cutoff frequency sync/ch1/ch2
  - direct level sync/ch1/ch2
  - direct phase sync/ch1/ch2
  - effect level sync/ch1/ch2
  - effect phase sync/ch1/ch2
  - effect modulation sync/ch1/ch2
  - effect mod phase sync/ch1/ch2
  - QUIT
- ALG20:RTY
  - rotary speaker on/off
  - frequency scale
  - speed select
  - horn speed (fast)
  - horn speed (slow)
  - horn speed modulation
  - horn speed mod phase
  - rotor speed (fast)
  - rotor speed (slow)
  - rotor speed modulation
  - rotor speed mod phase
  - rising time (horn)
  - falling time (horn)
  - rising time (rotor)
  - falling time (rotor)
  - horn depth
  - rotor depth
  - drive
  - level
  - ambience
  - horn : rotor mix balance
  - output level
  - QUIT

BLOCK	ALGORITHM	PARAMETER

30 ▽

- PRE-EF1&2
  - ALG0:OFF
    - QUIT
  - ALG1:SEQ
    - stereo EQ on/off
    - bass frequency sync/ch1/ch2
    - bass level sync/ch1/ch2
    - treble frequency sync/ch1/ch2
    - treble level sync/ch1/ch2
    - PEQ frequency sync/ch1/ch2
    - PEQ level sync/ch1/ch2
    - PEQ q sync/ch1/ch2
    - QUIT
  - ALG2:SXE
    - stereo exciter + EQ on/off
    - exciter level sync/ch1/ch2
    - bass frequency sync/ch1/ch2
    - bass level sync/ch1/ch2
    - treble frequency sync/ch1/ch2
    - treble level sync/ch1/ch2
    - QUIT
  - ALG3:DEX
    - dynamic exciter on/off
    - trigger select
    - exciter sensitivity
    - exciter type
    - release time
    - QUIT
  - ALG4:GTE
    - gate on/off
    - trigger select
    - attack time
    - release time
    - threshold level
    - hysterisis level
    - predelay time sync/ch1/ch2
    - QUIT
  - ALG5:CMP
    - compressor on/off
    - trigger select
    - attack time
    - release time
    - threshold level
    - ratio
    - gain make up
    - predelay time sync/ch1/ch2
    - QUIT

31 ▽

Classification Chart for Editing

BLOCK	ALGORITHM	PARAMETER

(31)

- POST-EF
  - ALG0:OFF
    - QUIT
  - ALG1:SEQ
    - stereo EQ on/off
    - bass frequency sync/ch1/ch2
    - bass level sync/ch1/ch2
    - treble frequency sync/ch1/ch2
    - treble level sync/ch1/ch2
    - PEQ frequency sync/ch1/ch2
    - PEQ level sync/ch1/ch2
    - PEQ q sync/ch1/ch2
    - QUIT
  - ALG2:SXE
    - stereo exciter + EQ on/off
    - exciter level sync/ch1/ch2
    - bass frequency sync/ch1/ch2
    - bass level sync/ch1/ch2
    - treble frequency sync/ch1/ch2
    - treble level sync/ch1/ch2
    - QUIT
  - ALG3:DEX
    - dynamic exciter on/off
    - trigger select
    - exciter sensitivity
    - exciter type
    - release time
    - QUIT
  - ALG4:GTE
    - gate on/off
    - trigger select
    - attack time
    - release time
    - threshold level
    - hysterisis level
    - predelay time sync/ch1/ch2
    - QUIT
- ENV
  - ALG0:OFF
    - QUIT
  - ALG1:EVF
    - envelope follower on/off
    - signal select
    - input multiplier
    - detector attack time
    - detector release time
    - QUIT
  - ALG2:EG1
    - envelope generator 1 on/off
    - trigger select
    - threshold level
    - trigger mask time
    - trigger on delay time
    - attack time
    - hold time
    - decay time
    - sustain time
    - release time
    - start level
    - attack level
    - sustain level
    - attack curve
    - decay curve
    - release curve
    - detector attack time
    - detector release time
    - QUIT

(32) (33)

(to be continued)

BLOCK	ALGORITHM	PARAMETER

(32) (33)

- ALG3:EG2
  - envelope genarator 2 on/off
  - trigger select
  - threshold level
  - hysterisis level
  - trigger mask time
  - trigger on delay time
  - trigger off delay time
  - attack time
  - hold time
  - decay time
  - release time
  - start level
  - attack level
  - sustain level
  - attack curve
  - decay curve
  - release curve
  - detector attack time
  - detector release time
  - QUIT

- INPUT
  - input level sync/ch1/ch2
  - input phase sync/ch1/ch2
  - input panpot sync/ch1/ch2
  - QUIT
- OUTPUT
  - output level sync/ch1/ch2
  - output phase sync/ch1/ch2
  - output panpot sync/ch1/ch2
  - QUIT
- LCL.MIDI
  - control no.1
  - parameter block 1
  - parameter name 1
  - parameter range 1
  - control no.2
  - parameter block 2
  - parameter name 2
  - parameter range 2
  - control no.3
  - parameter block 3
  - parameter name 3
  - parameter range 3
  - control no.4
  - parameter block 4
  - parameter name 4
  - parameter range 4
  - QUIT
- SYSTEM
  - input mode
  - auto help
  - load form
  - load time
  - dial sensitivity
  - unit (time)
  - unit (level)
  - unit (q)
  - remote ch
  - remote baud rate
  - clock set
  - user's name
  - date of birth
  - key protect
  - key trigger
  - battery check
  - version check
  - QUIT

(34)

(to be continued)

BLOCK	ALGORITHM	PARAMETER

34

- MEMORY
  - memory compare
  - move
  - copy
  - delete
  - exchange
  - remaining area
  - protect U1
  - :
  - protect U256
  - QUIT
- SYS.MIDI
  - MIDI on/off
  - MIDI omni
  - MIDI ch
  - bulk dump transfer
  - bulk dump receive on/off
  - program change no.1
  - :
  - program change no.128
  - QUIT
- B.LOAD
  - MOD block load
  - PRE-EF1 block load
  - PRE-EF2 block load
  - POST-EF block load
  - ENV block load
  - QUIT

Specifications

A/D converter	18 bit oversampling stereo A/D converter
D/A converter	Advanced pulse D/A converter
Sampling frequency	48 kHz

Input

Connector type	Reference input level	Max. input level	Input impedance	Circuitry type
XLR-3-31 equivalent	+4 dBs	+24 dBs	10 kilohms	Balanced
Phone jack	– 10 dBs	+10 dBs	50 kilohms	Unbalanced

XLR-3-31 equivalent connector (1: GND 2: HOT 3: COLD)

Output

Connector type	Reference output level	Max. output level	Suitable load impedance	Circuitry type
XLR-3-32 equivalent	+4 dBs	+24 dBs	Over 600 ohms	Balanced
Phone jack	–10 dBs	+10 dBs	Over 10 kilohms	Unbalanced

XLR-3-32 equivalent connector (1: GND 2: HOT 3: COLD)

Frequency characteristics
10 Hz — 22 kHz $^{+0}_{-1.0}$ dB

S/N	Over 97 dB
Dynamic range	Over 97 dB
Distortion rate	Under 0.0035% (at 1 kHz)
Memory	
Preset memory	100 types
User memory	Max. 256 types
Power requirement	USA and Canadian model 120 V AC, 60 Hz UK model 240 V AC, 50/60 Hz (adjustable with a voltage selector) Continental European model 230 V AC, 50/60 Hz (adjustable with a voltage selector)
Power consumption	27 W
Dimensions	Approx. 482 X 44 X 320 mm (w/h/d) (19 x 1 3/4 x 12 5/8 inches) (excluding projections)
Weight	5.0 kg (11 lb 1 oz)
Accessories supplied	Power cord (1) Preset memory list (1)

Design and specifications are subject to change without notice.

Note:
This appliance conforms with EEC Directive 87/308/EEC regarding interference suppression.

Troubleshooting

Symptom	Check if:
Power does not go on	• The power cord is plugged into the outlet.
No sound	• The INPUT control is set to "0". • The OUTPUT control is set to "0".
No sound effect	• The bypass circuitry is functioning.
Sound is distorted	• Input level is too high. → Lower the input level by turning the INPUT control counterclockwise.
No stereo effect	• "input mode" of the system block is set to "mono."
Uncontrollable with MIDI	• MIDI receive channel is suitable for sending channel of the MIDI device. • The control number assigned to this unit is used. • "MIDI on/off" of the SYS. MIDI block is set to "on."

Glossary

Parameter
A number of elements are involved in creating each effect. One effect is obtained only after determining the values of the elements required. Each of these elements is called a parameter.

Indirect parameter
This is a parameter that can be changed according to preset rules while editing. "scale" and "sync" are typical examples. This is not an actual parameter (parameter that can be saved) but is a convenient parameter that can be changed in multiple lots.

Memory
This is an internal memory circuit. The DPS-M7 has a built-in microcomputer that sends the set value of each parameter to the signal processing LSI (DSP) to create the various effects. If the data of this parameter is stored in the memory, it can be retrieved and used when needed.
The DPS-M7 has 100 preset memories (memory initially set at time of shipment) and a maximum of 256 user memory (memories that are available to the user).

Temporary buffer
This is a place where the parameter of each effect is loaded and edited. Each effect is reproduced by the parameters called into this temporary buffer.

Load
Calling the effects stored in the memory is called "to load." The parameters stored in the preset memory and user memory are copied in the temporary buffer and then new parameters are reflected in the DSP. Partial loading of the memory is executed in the B. LOAD block of the edit mode.

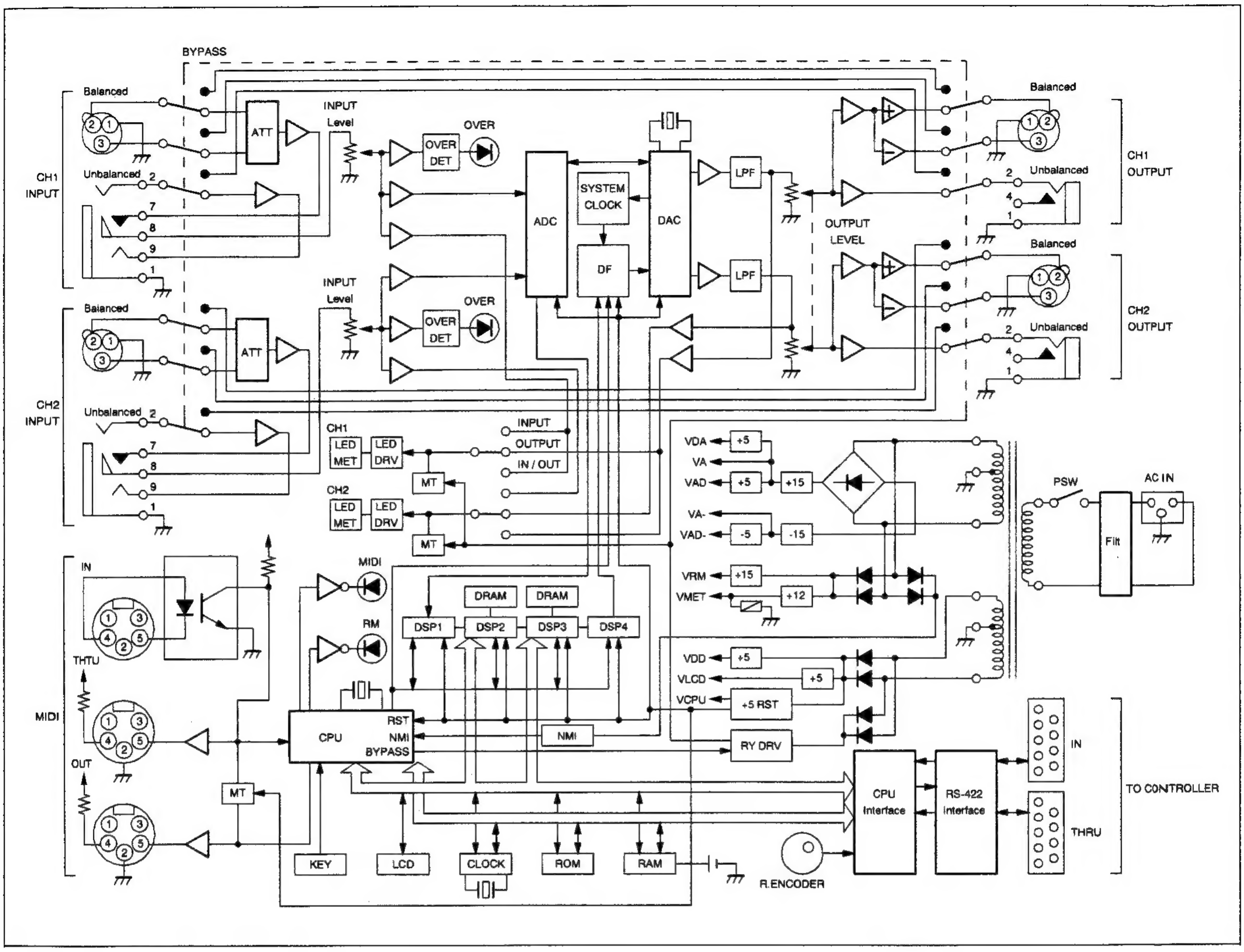
Edit
Changing the value of a parameter is called "to edit," and original effects can be created by changing values of parameters in the temporary buffer. This function is to make the effects in the preset memory more effective by conforming with usage conditions and the user's own tastes.

Save
Storing parameters in the temporary buffer as user memory is called "to save" and is an important function to store original effects. Original effects once saved can be freely accessed for editing and saving again.

MIDI
This is the abbreviation for Musical Instrument Digital Interface and is an international standard for data communication between electronic musical instruments. This enables automatic performance by controlling other musical instruments from one keyboard or by using a sequencer and computer. The MIDI function of the DPS-M7 enables selection of memory numbers with MIDI program change numbers (tone quality change signal from the keyboard) and control of parameters by means of the MIDI control change signal (amount of change of the modulation wheel and so on).

Algorithm
A fundamental arithmetic method is required in the internal circuit of the digital sonic modulator to obtain an effect and different arithmetic methods are used such as for chorus effects, pitch effects and flanger effects. Any one of these arithmetic methods is called an algorithm. Great many newly developed algorithms are incorporated in the DPS-M7 for variegated effects far exceeding those available from conventional effectors.

Block Diagram/Schéma fonctionnel



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